Tenor® AS VoIP MultiPath/Gateway Switch

Product Guide

P/N 480-0059-00-10

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About this Guide

What's included?

This product guide is divided into chapters; each chapter describes a specific topic. The following chapters are included:

- *About this Guide:* Describes what is included in the Product Guide, including typographical conventions.
- *Chapter 1: Overview.* Includes a general overview of the product, including a description of the *Tenor AS's* features and capabilities.
- *Chapter 2: Hardware Components.* Hardware description, including the front and rear panels, as well as LEDs and required cables.
- *Chapter 3: Installation/Basic Troubleshooting.* Describes how to install the *Tenor AS* unit, including how to connect, power up and assign the IP address.
- *Chapter 4: Advanced Topic: View Call Detail Records.* Describes the Call Detail Recording (CDR) feature, including how to set up the CDR server and assign a password. In addition, instructions for reading CDR output are also included.
- *Chapter 5: Advanced Topic: Diagnostic/Maintenance:* Describes how to troubleshoot and monitor the health of the system.
- *Chapter 6: Advanced Topic: SNMP/IVR:* This chapter describes the SNMP protocol and how to use it with the *Tenor AS.* In addition, it describes the Interactive Voice Response (IVR) system for support of pre-paid and post-paid calls.
- *Glossary*
- *Index*
- *Warranty*

Typographical Conventions

Product Guide Conventions

Certain typographical conventions are used throughout this product guide. See below.

- All commands you enter via keystrokes appear in **bold** (e.g., Press **Enter** or Press **Ctrl-I).**
- All text commands you enter via Telnet session or command line typing appear in *italics* (e.g., type *active).*
- There are three types of special text that are designed to reveal supplemental information: Note, Warning, and Caution. See below.

A **NOTE** provides additional, helpful information. This information may tell you how to do a certain task or just be a reminder for how-to's given in previous sections. (i.e., For a list of valid commands at any time, type *?*)

A **WARNING** provides information about how to avoid harming your VoIP equipment or other equipment (i.e., Do not stack more than 4 units together.)

A **CAUTION** provides information about how to avoid injury to yourself or to others (e.g., Do not install the equipment during a lightning storm).

Finding Help

Refer to the Product Guide for help. The Table of Contents and Index tells you where to find information easily.

Extensive configuration help is available from the *Tenor Configuration Manager/Tenor Monitor* User Guide or the *Command Line Interface* User Guide. Both documents are on the CDR ROM you received with unit or you can download the latest documentation from **www.quintum.com**

Chapter 1: Overview

This chapter gives you a general overview of the *Tenor AS* including feature descriptions and capabilities. Specifically, the following topics are covered:

- *A description of Tenor AS*
- *Features*
- *Capabilities*
- *Call Paths*
- *H.323 Gatekeeper Services*
- *Advanced Features*

What is the *Tenor AS?*

The *Tenor AS* is a VoIP (Voice over Internet Protocol) H.323/SIP switch that digitizes voice, fax, and modem data and transmits it over the IP network. Designed as a SOHO product, the *Tenor AS* gives small to medium sized businesses with analog voice infrastructure an easy, cost-effective way to capitalize on the power of Voice over IP (VoIP).

The *Tenor AS* integrates a gateway, gatekeeper, border element intelligent call routing, and supports H.32/SIP and QoS all in one solution. The gateway converts circuit switched calls to VoIP calls, the gatekeeper performs IP call routing functions, and the border element distributes the call routing directories throughout the network. Through the FXS port, you can connect a telephone, key system or PBX; through the FXO port, you can connect to the PSTN (through direct connection to the Central Office).

Figure 1-1 Tenor AS VoIP Switch

The *Tenor AS* is available in two configuration types:

- **ASG VoIP Gateway.** The *ASG VoIP Gateway* is mainly intended for applications interfacing between the PBX and the VoIP network. The number of VoIP channels equals the number of FXS ports. Calls can be routed in any direction between any of the ports.
- **ASM Multipath Switch.** The AXM MultiPath Switch is mainly intended for symmetrical multipath applications, with an equal number of FXO and FXS ports.The number of VoIP channels is equal to half the number of telephony (i.e. PSTN) channels.

  With its MultiPath architecture, the *Tenor AS* can intelligently route calls between the FXS, FXO, and the VoIP network to achieve the best combination of cost and quality. The *Tenor AS* also routes calls over IP to reduce costs, and then transparently “hops off” to the PSTN, to reach off-net locations. Calls can be routed in any direction between any of the ports.

Table 1-1 Tenor AS Configuration Types

Series	Configuration	FXS Ports	FXO Ports	VoIP Ports
ASM MultiPath	ASM200	2	2	2
	ASM400	4	4	4
ASG VoIP Gateways	ASG200	2	0	2
	ASG400	4	0	4

Whichever configuration you choose, the high performance unit provides one Autosensing BaseT connection, along with one RS-232 serial console port connection. The unit also incorporates an intelligent call routing engine which regulates system resources and configuration while coordinating all voice traffic activity in the unit.

The unit's simple plug-and-play embedded system architecture brings VoIP technology to your network without changing your existing telephony infrastructure. Your network stays as is, and the call type is transparent to the user. This technology boasts superior voice quality without compromising reliability.

Features

The *Tenor AS*'s specific features are explained below.

Unique Design

Tenor AS packs powerful VoIP features into one compact unit. The Tenor can be installed without upgrades to the existing voice or data network. You can install the unit in a home or office environment, without affecting the network infrastructure you already have in place. As with all Tenor architecture, the Tenor AS provides the power of VoIP in an easy-to-use product that takes just minutes to get up and running.

State-of-the-Art GUI Configuration and Network Management

The *Tenor AS* is managed/monitored by the *Tenor Configuration Manager* and *Tenor Monitor*. Through the *Tenor Configuration Manager*, you can configure all options, such as dial plans, call routing numbers, etc. An easy-to-use Java-based installation process enables you to an install the manager and start configuring within minutes. Through the *Tenor Monitor*, you can monitor the health of the system, including alarms, call detail records, etc. Both the *Tenor Configuration Manager* and *Tenor Monitor* provide comprehensive on-line help systems that are available at your fingertips.

In addition, you can configure the unit via *Command Line Interface (CLI*). Through this telnet session, you can access all configuration options, including an online help system, built into the CLI, which provides help for all features and functions. Just type *help* at any prompt, and data about that field will be displayed.

Easy Connect to Console

Plugging a serial cable between the unit's RS-232 port and your PC's console port, will allow local unit management. Through the console connection, you are able to assign an IP address. In addition, through the RS-232 port, you are able to configure the unit via *Command Line Interface (CLI).*

Powerful System Monitoring

There are many different ways to monitor the health of the unit, including LEDs and alarms. LEDs appear on the front of the unit.The LEDs light up according to operations and alarms the system is experiencing.

For more advanced monitoring, see *Chapter 5: Advanced Topic: Diagnostics/Maintenance* to view a list of active alarms, as well as view an alarm history. Each alarm indicates the unit's operational status.

Capabilities

SelectNet™ Technology Safety Net

Quality of service is virtually guaranteed. *Tenor AS* 's built-in patented SelectNet™ Technology provides a "safety net," which virtually guarantees that each call going VoIP will not only be routed successfully, but will deliver high voice quality.

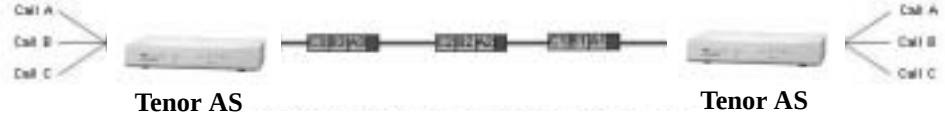
SelectNet monitors the IP network performance for VoIP calls. If the performance characteristics become unacceptable—according to the delay, jitter, and packet loss specifications you configure—the *Tenor AS* will switch the call to the PSTN automatically and transparently. The Tenor continuously monitors your data network for jitter, latency and packet loss, and transparently switches customer calls to the PSTN when required.

PacketSaver™ reduces bandwidth consumption

PacketSaver packet multiplexing technology reduces the amount of IP bandwidth required to support multiple calls flowing between two endpoints. PacketSaver minimizes bandwidth usage by aggregating packet samples from multiple VoIP conversations and packing them into a larger IP packet with a single IP header. The process removes the need to send a bulky IP header with individual voice packets. As a result, it eliminates the transmission of redundant information.

Figure 1-2 PacketSaver

Conventional VoIP Transmission Sends Many Redundant Packet Headers



Tenor using PacketSaver to Minimize Bandwidth Usage

Virtual Tie Trunk

The Tenor unit can emulate any tie trunk. It provides all of the functionality of a tie trunk, including the considerable cost savings, but eliminates the need for a PBX trunk to be configured, or marked as a tie trunk. (A traditional tie trunk is a PBX-configured direct connection between two PBXs in separate locations. The tie trunk bypasses the PSTN network, which results in considerable savings.)

Your PBX does not need any additional configuration. The *Tenor AS* treats all trunks the same without compromising voice quality.

SNMP Support

The *Tenor AS* supports Simple Network Management Protocol (SNMP), the standard protocol used to exchange network information between different types of networks. The *Tenor AS* unit acts as an SNMP agent—using HP® Openview™—to receive commands and issue responses to the Network Manager. The Network Manager will then be able to perform certain functions, such as receiving traps from the *Tenor AS*.

Call Detail Recording

Through the Call Detail Record (CDR) feature, the *Tenor AS* generates a call record at the completion of each call, typically for accounting purposes. A CDR is a string of data that contains call information such as call date and time, call duration, calling party, and called party. *Tenor AS* may store Call Detail Records locally or they can be sent to a CDR server within the network. The CDR contains sufficient information to capture billing data, which can be used to create billing reports using third party billing software.

IVR/RADIUS Support

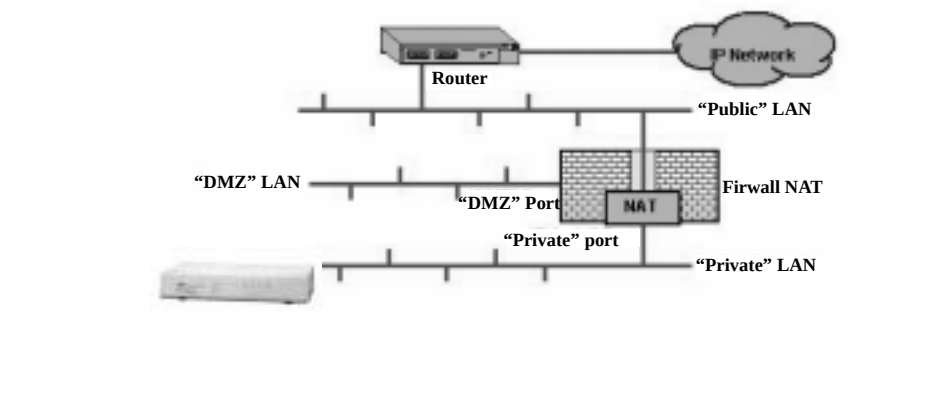
Interactive Voice Response (IVR) is a feature of the *Tenor AS* that enables you to offer services, such as Pre-paid calling cards and Post-paid accounts, to your customers.

The Tenor uses the RADIUS (Remote Authentication Dial-In User Service), for authenticating and authorizing user access to the VoIP network, including ANI Authentication (Types 1 and 2). The RADIUS is a standard protocol which provides a series of standardized message formats for transmitting and receiving dialed information, account data and authorization codes between the network access gateway and the billing server.

NAT*Access*™

NAT*Access* is an intelligent network address translation technology. It enables VoIP networks with multiple H.323 endpoints to operate behind firewalls equipped with H.323 Network Address Translation (NAT); this provides maximum network security. NAT*Access* simplifies deployment by eliminating the need to place the Tenor on a public IP network. Using NAT*Access* provides easy, secure expansion between mulitple VoIP sites. In addition, NAT technology in the Tenor permits the use of private subnets at the same time; in-house calls will never go over the public internet.

Figure 1-3 Tenor with NAT*Access* Deployment



Dynamic Call Routing

Tenor AS's intelligent call routing capabilities are state-of-the-art. The unit automatically detects and supports three call types: voice, fax, and modem.

Tenor AS will first identify the call origination site—Line/FXO, Phone/FXS, or IP routing group — and then route the call according to the parameters you have configured in the routing database. Each call may be routed via circuit switched path between any two circuit groups, or compressed and transported via VoIP when connecting to an IP routing group. Trunk circuits are those that typically connect to another circuit switched network such as the PSTN. Line circuits typically connect to a termination device on the user premises, such as a PBX.

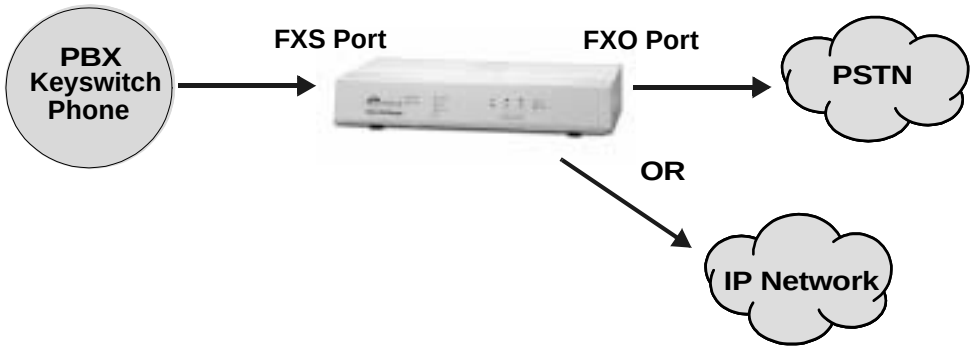
Tenor AS Call Paths

Tenor MultiPath Switch (ASM200 and ASM400) Configuration

The Tenor MultiPath Switch Configuration is symmetrical with an equal number of Phone/FXS and Line/FXO ports. Calls are routed from the Phone/FXS, Line/FXO, or IP Network. Calls can be routed in any direction between any of the ports.

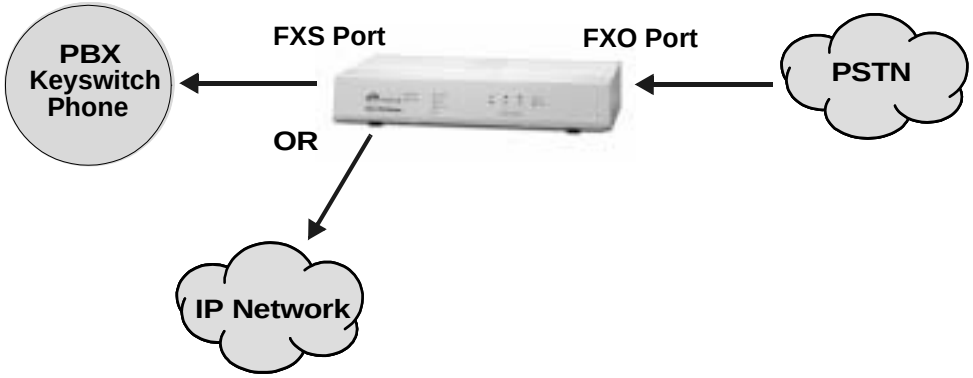
FXS (Phone) Originated Calls. Calls coming from the Phone/FXS interface (i.e. PBX) may be switched to either the data network as a VoIP call or to the FXO interface, typically for connection to another circuit switched network such as the PSTN. The routing decision made by the Tenor is based upon your configuration and the dialed number. See Figure 1-4 for an example of a call originated from a PBX.

Figure 1-4 FXS (Phone) Originated Calls



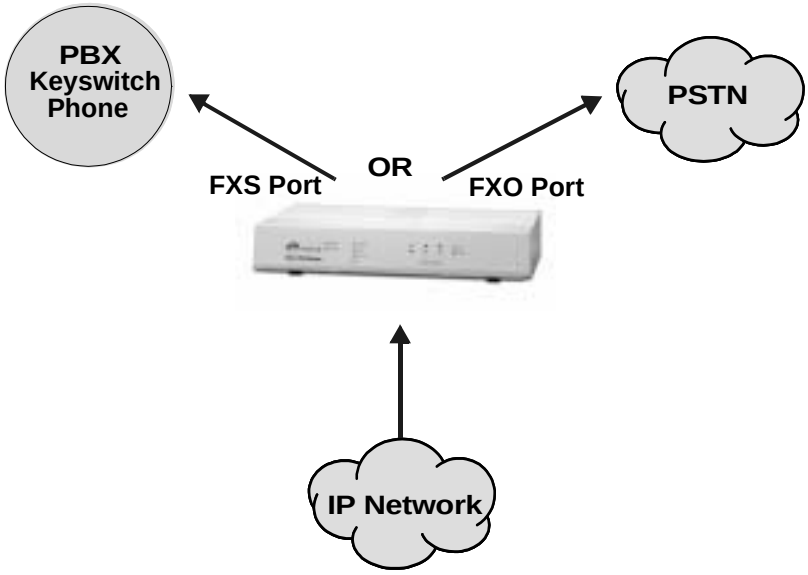
FXO (Line) Originated Calls. A call coming from a Line/FXO interface may be switched to either the data network as a VoIP call, a Line Circuit, or trunk typically for connection to a termination device on the user's premises such as a PBX. The routing decision made by the *Tenor AS* is based upon your configuration and the dialed number. See Figure 1-5 for an example of a call originated from the PSTN.

Figure 1-5 FXO (Line) Originated Calls



IP Network Calls. Calls coming from the IP network (data network) can be routed to the Line/FXO or Phone/FXS interfaces. The Tenor will route calls based upon the dialed number. If the number is configured as a local phone number, the call will be sent to a Phone/FXS circuit for termination, otherwise the call is considered a “Hop-Off call” and the Tenor sends it out through a Line/FXO interface, typically connected to the PSTN. See Figure 1-6 for an example of a call originated from the IP network.

Figure 1-6 IP Network Originated Calls

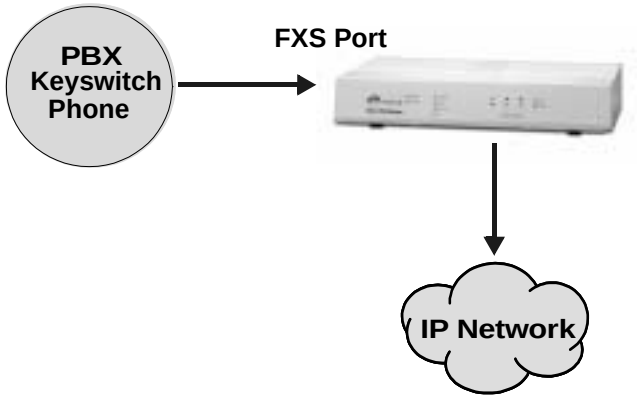


Tenor VoIP Gateway (ASG200 and ASG400) Configuration

The Tenor VoIP Gateway Configuration is used for Phone/FXS port connecting to the VoIP Network. The number of VoIP ports is equal to the number of FXS ports. Calls can be routed in any direction between any of the ports. See Figure 1-7 for an example of a call origination from the PBX.

FXS (Phone) Originated Calls. Calls coming from the Phone/FXS interface (i.e. PBX) may be switched to the data network as a VoIP call. The routing decision made by the Tenor is based upon your configuration and the dialed number. See Figure 1-7 for an example of a call originated from a PBX.

Figure 1-7 FXS (Phone) Originated Calls



Advanced Features/Capabilities

Call Management

There are four types of routing databases you can configure: Bypass Directory Numbers (BPN), Hunt Local Directory Numbers (Hunt LDN), Hop-Off Directory Numbers (HDN) and Static Routes.

Bypass Directory Numbers. Bypass Directory Numbers (BDN) are telephone numbers that are automatically routed directly from a line circuit to the PSTN; they will not be routed VoIP. Some examples of bypass numbers include toll-free calls, emergency calls (i.e., 911), or high security calls.

Hunt Local Directory Numbers. A Hunt Local Directory Number ((Hunt LDN) is a phone number reachable through local Line Circuits.

Hop-Off Directory Number. A Hop-off PBX call travels over IP, and then "hops" off into the public network (PSTN) on the destination side to reduce or eliminate public toll charges (also known as Leaky Area Map). A Hop-Off Directory Number is routed over the IP to another Tenor location and then out to the Trunk circuit, possibly to the PSTN as a local call.

Static Routes. Static Routes are used between networks and other H.323 devices that are not registered to the network through the Border Element (such as non-Quintum gateways). A static route associates endpoints (as represented by their IP address) with Directory Number patterns.

Dial Plan Options

Public/Private Dial Plan Support. The *Tenor AS* supports public and private dial plans. A public dial plan includes numbers which conform to the international dialing plan (E.164) of a country code + city/area code + local number. For a public dial plan, you can define the numbering plan structure for the *Tenor AS* to use for outgoing calls.

A private dial plan does not conform to a public dialing plan (i.e., 3 digit dialing plan); through the *Tenor AS* you are able to configure the unique pattern/dialing plan structure, including number length.

You are able to configure which dial plan to use for incoming and outgoing calls, including whether other options such as hop-off calls, will use a public or private dial plan.

User Programmable Dial Plan Support. The User Programmable Dial Plan Support (UPDP) enables the Tenor to identify a completely customizable set of digit sequences, such as Local, National, International or Private Numbers.

PassThrough support for certain call types. Certain call types can be directly routed to a trunk circuit, without going IP. There are several routing tables you can configure via the *Tenor Configuration Manager* to adjust how the *Tenor AS* unit routes these types of "pass through" numbers. For example, you may want to configure 911 as a "bypass number", which means that all 911 calls coming into *Tenor AS* from the line circuit will be routed directly to a Trunk circuit presumably connected to a PSTN. Bypass calls are never routed over IP.

Hop-off PBX Calls. Hop-off numbers are phone number patterns for calls to be routed out to the PSTN. (A hop-off PBX call is a toll call which hops through a private network to reduce or eliminate the toll charge.) They are entered in a Hop-off Number Directory and associated with trunks where matching calls should be sent.

Tenor AS supports those hop-off PBX calls where the destination *Tenor AS* is programmed to route the call to the PSTN. The destination *Tenor AS* unit is configured with the phone numbers to be "supported" for this feature.

H.323 Gatekeeper Services

The *Tenor AS* unit's built-in H.323 gatekeeper performs IP call routing functions, such as call control and administrative services to another *Tenor AS* unit, or another H.323 endpoint. The gatekeeper's functionality complies with the H.323 industry specifications for voice control and management.

Gatekeeper. A Gatekeeper in an H.323 network provides call control services and other services to H.323 endpoints (i.e., gateways, terminals, and MCUs). The *Tenor AS* has a built-in H.323 gatekeeper which complies to the H.323 industry specifications for voice control and management. The gatekeeper performs call routing functions for calls entering and exiting a site.

The Gatekeeper performs IP call routing functions, such as Call Control Signaling and Call Authorization for Gateways, IP phones, and H.323 terminals. The Gatekeeper communicates with other Gatekeepers through a Border Element. When using a group of *Tenor AS* units, you can assign one unit as the Gatekeeper for the network. We recommend you configure each as its own gatekeeper.

Tenor AS supports gatekeeper to gatekeeper communication using the standard LRQ (Location Request)/LCF (Location Confirm) messaging scheme.

Zone Management. A zone is a group of H.323 defined endpoints controlled by a Gatekeeper. Endpoints can be gateways (i.e., *Tenor AS*), terminals, and/or multipoint conferencing units (MCUs). Endpoints establish control channels with a gatekeeper for registration, admission, and security. Call routing information about the endpoint is sent to the gatekeeper, including: IP address, unit type (gateway, terminal, or MCU) and routing information (such as phone numbers, number patterns, etc.).

A collection of zones is an administrative domain. An administrative domain provides call routing services for its zones through gatekeeper to gatekeeper messages or gatekeeper to border element messages (see below for more information).

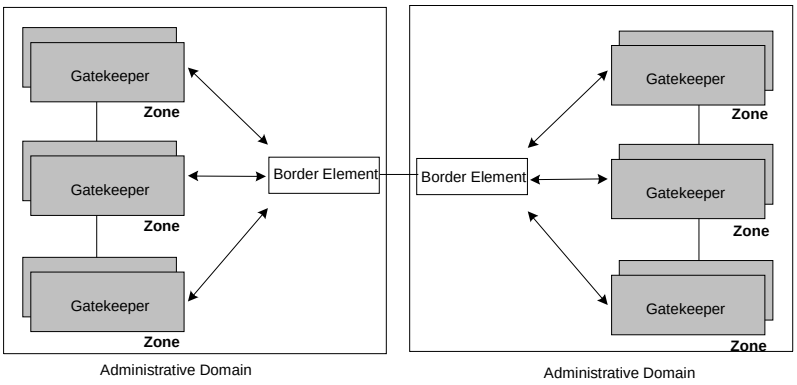
Call Registration. When registration from an H.323 endpoint is complete and a call is originated, the call request is sent to the gatekeeper. The call request provides the Gatekeeper with the dialed number and requests the routing information. The gatekeeper confirms the dialed number and supplies the endpoint with the destination IP address. For example, a *Tenor AS*'s gatekeeper will act as the gatekeeper for that zone and all of the other endpoints will register with it.

Border Element. The *Tenor AS*'s gatekeeper uses a border element to gain access to the routing database of the administrative domain for the purpose of call completion or any other services that involve communications with other endpoints out of the administrative domain. The border element functionality is built into the *Tenor AS* unit, along with the gateway and gatekeeper.

The primary function of the border element is to collect, manage, and distribute call routing information. A gatekeeper will establish a service relationship with a border element; the gatekeeper provides its zones capabilities and the border element shares call routing capabilities of other zones in the administrative domain. Through the border element, gatekeepers from multiple zones will be able to communicate.

A border element also establishes relationships with other border elements to route between administrative domains. If a gatekeeper cannot resolve an address, it contacts the border element.

In addition, if you are using more than one Tenor unit, you can configure one of the border elements for that zone. The *Tenor AS* unit can use two border elements: primary and secondary. These work together as one entity to provide redundancy and fault tolerance; there are no hierarchal differences.



Call Services. Gatekeepers provide services such as addressing, authorization and authentication of terminals and gateways, bandwidth management, accounting, billing, and charging. Gatekeepers also provide call-routing services. Specifically, the *Tenor AS* Gatekeeper provides the functions which follow:

Address Translation. The gatekeeper translates telephone numbers into IP addresses and vice versa. It performs Alias Address (phone number) to Transport Address (IP address) translation when an endpoint requests service. The Gatekeeper uses a translation table to translate an Alias Address (an address such as an H.323 identifier that a user may not understand) to a transport address. The translation table is updated using Registration messages.

Autodiscovery. The gatekeeper is discovered in one of the following ways: An endpoint sends an IP broadcast called a Gatekeeper Request message (GRQ) message (which includes that correct gatekeeper name) to discover a Gatekeeper OR the endpoint will discover a gatekeeper by its IP address.

Routing. The gatekeeper identifies the IP address of endpoints in its administrative domain. The gatekeeper builds a routing database from information obtained from the border element and also from gateways and H.323 endpoints.

Admissions Control. All H.323 endpoints must register and request permission to enter the gatekeeper’s zone; the gatekeeper will confirm or deny access to the network. The gatekeeper authorizes

network access and protects the integrity of the network using Admission Request (ARQ), Admission Confirmation (ACF) and Admission Reject (ARJ) messages.

SIP User Agent

SIP (Session Initiation Protocol) is a signaling protocol used to establish a session on an IP network for voice control and management; it is a request-response protocol that closely resembles Hypertext Transfer Protocol (HTTP), which forms the basis of the World Wide Web. SIP re-uses many of the constructs and concepts of Internet protocols such as HTTP and Simple Mail Transfer Protocol (SMTP). The purpose of SIP is only to establish/change/terminate sessions. SIP is not concerned with the content or details of the session.

SIP is Transport layer-independent, which means it can be used with any transport protocol: UDP, TCP, ATM, etc. It is text-based, so it requires no encoding/decoding like H.323. And SIP supports user mobility, using proxies and redirecting requests to your current location.

When configured for SIP the Tenor will act as a SIP User Agent (Endpoint) as defined in IETF RFC3261. Multiple user agents allow for separate agents to be allocated to each SIP call. It will be able to gateway calls to and from the IP network, and Customer Premise Equipment (CPE) such as phones, PBX's, and FAX machines, or the Public Switched Telephone Network (PSTN). The Tenor SIP User Agent will work in conjunction with an external SIP proxy or redirect server to route and connect calls over SIP based networks.

There are three basic components of SIP:

1. User Agent (Endpoint)
   - client element, initiates calls
   - server element, answers calls
2. Network Server (Proxy Server or Redirect Server)
   - name resolution
   - user location
   - redirect and forking
3. Registrar
   - Stores registration information in a location service using a non-SIP protocol.

Chapter 2: Hardware Components

This chapter tells you what is contained in your hardware package. A description of each component is also included.

Specifically, the following topics are covered:

- *Hardware Description*
- *Cables*
- *Specification*

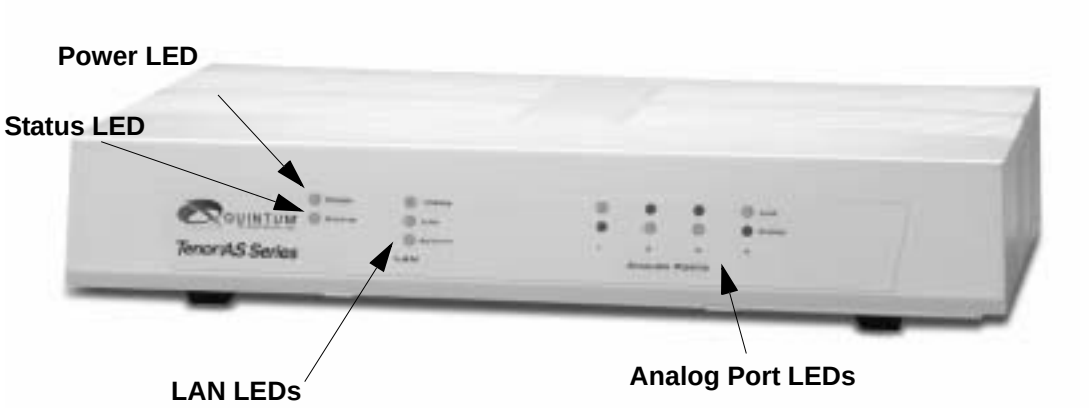
Hardware Description

The *Tenor AS* is a stackable device which provides Phone/FXS and Line/FXO connections as well as connections to the Ethernet LAN and a PC.

The unit's front panel includes LEDs; the back panel includes connection jacks, a diagnostics option, reset button, and an on/off power switch.

Front Panel Connections and Reset Options

Figure 2-1 Tenor AS Front Panel



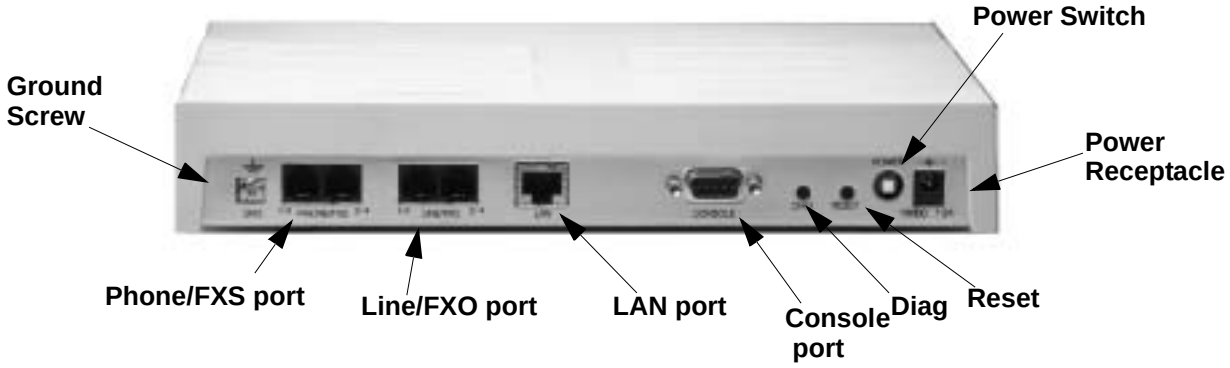
The LEDs display the health of the system. There are different types of LEDs: Power, Status, LAN and Analog Ports. A description of each is described in Table 2-1.

Table 2-1 Front Panel LEDs Definitions

LED	Label	LED Color	Description
Power	Power	Green	On: Indicates power is on. Off: Power is off.
Status	Status	Green Flashing	Operational Status. Off: *Tenor AS* is working properly. On: One or more diagnostic tests have failed.
Analog Ports	1, 2 3, and 4	Line LED - Green	On indicates activity is occurring on the Line/FXO port.
		Phone LED - Green	On indicates activity is occurring on the Phone/FXS port.

LED	Label	LED Color	Description
LAN	Link	Green	On: Link is working properly and there is activity on the line. Off: Link has failed.
	100Mb	Green	On: The advertised link rate is 100Mb if the link is not connected, or the actual link rate is 100b if the link is connected. Off: The advertised link rate is 10Mb if the link is not connected, or the actual link rate is 10Mb if the link is connected.
	Activity	Green Flashing	On: indicates there is activity (i.e., transmit/receive) on the line. Off: No activity is occurring.

Back Panel



- **Ground Screw.** An earth ground screw is provided to connect to supplemental earth ground using a Ground Safety Cable, if supplemental ground is needed.
- **Phone/FXS port.** Provides an RJ-11 jack for connection to a PBX, Keyphone or phone.

Figure 2-2 RJ-11 Phone/FXS Port Pin Order

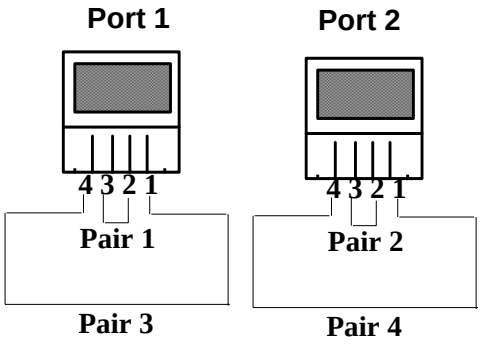


Table 2-2 Input/Output RJ-11 Phone/FXS port

	Pin #	Used with	Definition
Port 1	1	Port 1, pin 4	Tip
	2	Port 1, pin 3	Ring
	3	Port 1, pin 2	Tip
	4	Port 1, pin 1	Ring

	Pin #	Used with	Definition
Port 2	1	Port 2, pin 4	Tip
	2	Port 2, pin 3	Ring
	3	Port 2, pin 2	Tip
	4	Port 2, pin 1	Ring

- **Line/FXO port.** Provides an RJ-11 jack for connection to the Central Office to provide direct connection to the PSTN.

Figure 2-3 RJ-11 Phone/FXO Port Pin Order

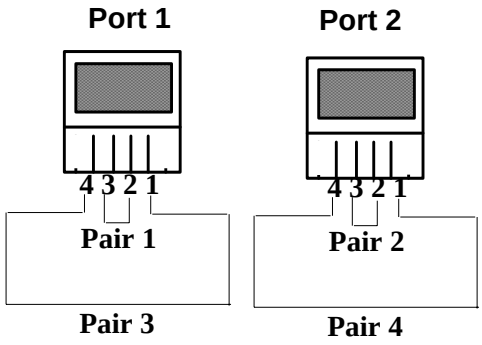


Table 2-3 Input/Output RJ-11 Phone/FXS port

	Pin #	Used with	Definition
Port 1	1	Port 1, pin 4	Tip
	2	Port 1, pin 3	Ring
	3	Port 1, pin 2	Tip
	4	Port 1, pin 1	Ring
Port 2	1	Port 2, pin 4	Tip
	2	Port 2, pin 3	Ring
	3	Port 2, pin 2	Tip
	4	Port 2, pin 1	Ring

- **LAN port.** 10/100 Base-T Ethernet port. This port provides an RJ-45 jack for an individual connection to a 10/100 Ethernet LAN switch or hub via RJ-45 cable; it is individually configured with a unique IP and MAC address.

Figure 2-4 10/100 BASE-T Ethernet Port Pin Order

Table 2-4 Input/Output 10/100 Ethernet port

Pin #	Signal	Definition	Color
1	TX +	Transmit Data	White w/orange
2	TX -	Transmit Data	Orange
3	RX +	Receive Data	White w/green
4	RSVD	Reserved	Blue
5	RSVD	Reserved	White w/blue
6	RX -	Receive Data	Green
7	RSVD	Reserved	White w/Brown
8	RSVD	Reserved	Brown

- **Console port.** This RS-232 connector is used for connection to a PC's serial port via DB-9 serial cable at 38400 BPS 8N1, without flow control. The input/output signals are listed in Table 2-5.

Figure 2-5 DB-9 Female Connector Pin Order

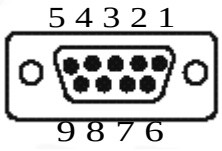


Table 2-5 Serial RS232 DB-9 Connector Pinouts

Pin #	Function	Description
1	DTR	Data Terminal Ready
2	TXD	Transmit Data
3	RXD	Receive Data
4	CD	Carrier Detect
5	GND	Signal Ground
6	N.C.	No Connect

Pin #	Function	Description
7	N.C.	No Connect
8	N.C.	No Connect
9	N.C.	No Connect

- **Diag.** Enables you to perform software diagnostic procedures.
- **Reset**. Enables you to reset the system. See *Chapter 5: Advanced Topic: Diagnostics/Maintenance* for more information.
- **Power Switch.** Switch to turn power on and off.
- **Power Socket.** Connection port to external power supply block.

Cables

The cables listed in Table 2-6 are required to connect a *Tenor AS* to various interfaces. Contact Quintum for ordering information, if necessary.

Table 2-6 Cables Supported

Cable	Usage
RJ-11 to RJ-11 Cable	Connection to FXO/Line. (For the AS400 unit. a Dual Port Splitter Cable is required.)
RJ-11 to RJ-11 Cable	Connection to FXS/Phone. (For the AS400 unit, a Dual Port Splitter Cable is required.)
RJ-11 Splitter Cable	Connection to ASM400 units enabled for four FX0 ports or to ASM400 units enabled for four FXS ports.
RJ-45 Ethernet cable	Connection to Ethernet LAN 10/100
DB-9 Serial RS-232	Connection to PC's asynchronous console port.
AC power adaptor/cord	One AC power adapter/cord for connection to AC power jack.

RJ-11 Cables (for ASM200, ASG200)

An RJ-11 cable is used to connect the *Tenor AS* to an FXO or FXS external device. The RJ-11 cable connector pinouts are given in this section to help you identify the proper connector to accommodate your specific networking requirements. The RJ-11 connector is the EIA/TIA standard for Unshielded Twisted Pair (UTP) cable; the wiring color codes are UTP Standard Coloring. The pin order is shown in Figure 2-6.

Figure 2-6 RJ-11 Pin Order

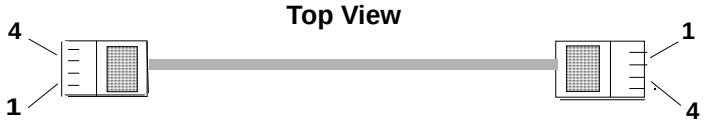


Figure 2-7 RJ-11 Connector Pinouts

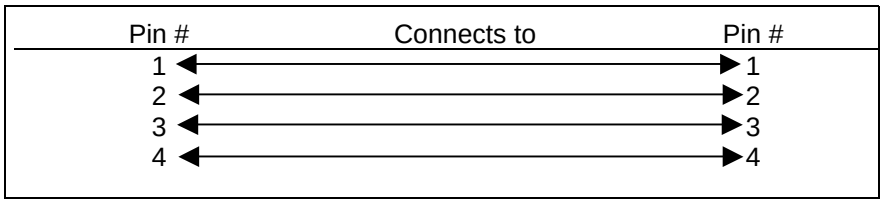


Table 2-7 RJ-11 Connector Pinouts

Pin #	Definition
1	TIP
2	Ring
3	Tip
4	Ring

RJ-11 Splitter Cable (for ASM400 and ASG400)

An RJ-11 cable is used to connect the *Tenor AS* to an FXO or FXS external device. The RJ-11 cable connector pinouts are given in this section to help you identify the proper connector to accommodate your specific networking requirements. The RJ-11 connector is the EIA/TIA standard for Unshielded Twisted Pair (UTP) cable; the wiring color codes are UTP Standard Coloring.

The RJ-11 Splitter cable's single end is labeled **Tenor** (see Figure 2-9); the split ends can be a 6-pin jack (accepts 4-pin plug or 6-pin plug) or 4-pin plug. Each end is labeled **A** or **B** (see Figure 2-9) for pin order.

For illustration purposes, the 4-pin plug is shown in Figure 2-8.

Figure 2-8 RJ-11 Splitter Cable

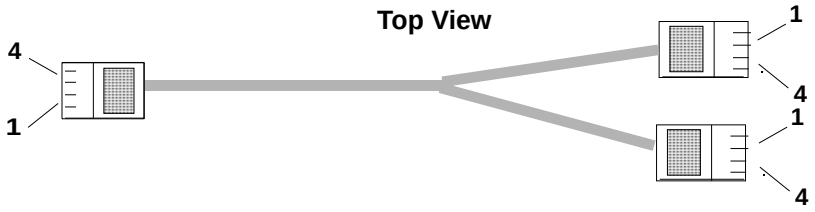
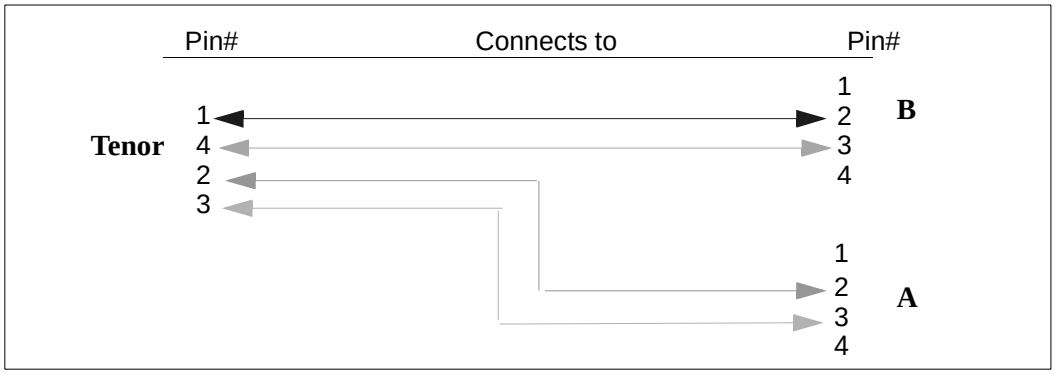


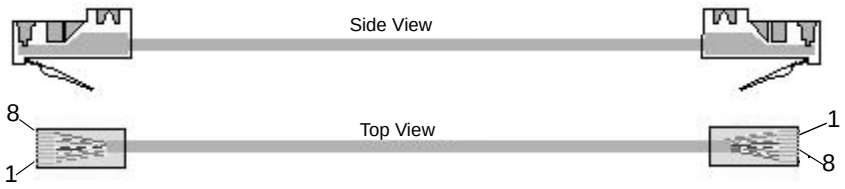
Figure 2-9 RJ-11 Splitter Cable Pin Order



RJ-45 Cables

RJ-45 cable connector pinouts are given in this section to help you identify the proper connector to accommodate your specific networking requirements. The RJ-45 (ISO 8877) connector is the EIA/TIA standard for Unshielded Twisted Pair (UTP) cable; the wiring color codes are UTP Standard Coloring. The pin order is shown in Figure 2-1.

Figure 2-1 RJ-45 Pin Order



RJ-45 Ethernet Cable (10/100)

An RJ-45 (10/100BaseT) straight through shielded cable is used to connect *Tenor AS* to an Ethernet LAN. Cable pinouts are listed in Table 2-1. Color specifications are applicable to the RJ-45 cable provided.

Figure 2-2 RJ-45 (10/100BT) Connector Pinouts

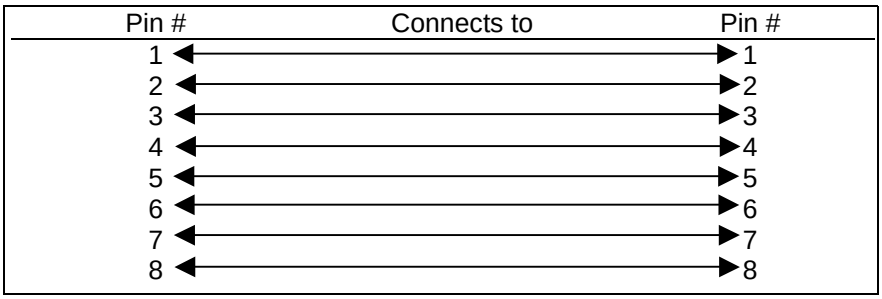


Table 2-1 RJ-45 (10/100BT) Connector Pinouts

Pin #	Signal	Definition	Color
1	TX +	Transmit Data	White w/orange
2	TX -	Transmit Data	Orange
3	RX +	Receive Data	White w/green
4	Unused	Unused	Blue
5	Unused	Unused	White w/blue
6	RX -	Receive Data	Green
7	Unused	Unused	White w/Brown
8	Unused	Unused	Brown

DB-9 Serial RS-232 Cable

The Serial RS-232 9-pin cable with a DB-9 male connector (with RS-232 interface) is used to connect the *Tenor AS* to your PC's asynchronous serial port. The pin order for DB-9 male and female connectors are shown in Figure 2-3 and Figure 2-4.

Figure 2-3 DB-9 Male Connector Pin Order

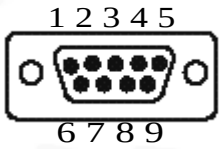


Figure 2-4 DB-9 Female Connector Pin Order

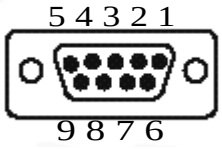


Figure 2-5 DB-9 Connector Pinouts

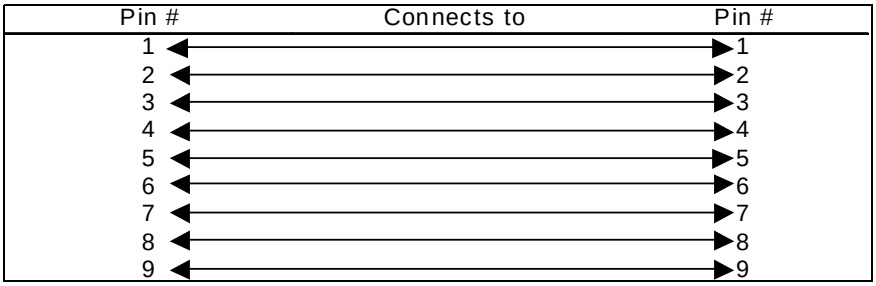


Table 2-2 DB-9 Connector Pinouts

Pin #	Function	Description	Pin #
1	DTR	Data Terminal Ready	1
2	TXD	TransmitData	2
3	RXD	Receive Data	3
4	CD	Carrier Detect	4
5	GND	Signal Ground	5
6	N.C.	No Connect	6
7	N.C.	No Connect	7
8	N.C.	No Connect	8
9	N.C.	No Connect	9

Specifications

Voice/Fax

Call Routing:	FXO/FXS/IP
Coding:	A-law, mu-law
Voice Algorithms:	G.723.1A (5.3, 6.3 Kbps), G.726 (16, 24, 32, 40 Kbps), G.729A, G711
Fax Support:	Group III at 2.4, 4.8, 7.2, 9.6, 12, 14.4 Kbps
Automatic Call Detection:	Voice/Modem/Fax

PSTN/PBX Connections

Interface:	Analog, FXO Interface (PSTN side), FXS Interface (PBX side)
Connector:	RJ-11
Ringing Cadence:	Adjustable, country-dependent
Maximum Loop Current:	24 mA
Ringer Equivalence Number:	3 up to 1000 feet of 24 AWG or heaver)
Ringing Frequency:	Adjustable, country-dependent
Ringing Voltage:	Adjustable, country-dependent

LAN Connection

LAN Support:	10/100 Mbps Ethernet
Connection Type:	Autosensing of speed and duplex

Physical

Depth:	7” (18.73 cm)
Height:	2” (5.1 cm)
Width	10 1/8 (24.5 cm)
Weight:	3.5 lbs (1.6 kg)

Electrical

Ethernet:	Standard 10/100Base-T RJ-45 interface (IEEE 802.3)
FXO/FXS:	Standard RJ-11 connectors
Connectors:	2 RJ-11 connectors for FXO connection 2 RJ-11 connectors for FXS connection
Console Port:	RS-232/DB-9 Female
Power:	12Vdc at 1.2A max. Wall adapter is supplied with each unit to derive the 12Vdc needed

Environmental

Operating Temperature:	40° to 104° F (5 - 40° C)
Operating Humidity:	10% to 90% non-condensing
Altitude:	-200 to 10,000 feet, or -60 to 3,000 meters
Storage Temperature:	14° to 140° F, or -10° to 60° C

Chapter 3: Installation/Basic Troubleshooting

This chapter gives you installation instructions, as well as how to position the *Tenor AS* successfully within your network.

Specifically, the following topics are covered:

- ❑ *Installation*
- ❑ *Install Ground Safety Cable*
- ❑ *Power up the System*
- ❑ *Assign IP Address*
- ❑ *Load Software Upgrade*
- ❑ *Getting Started with Configuration*
- ❑ *Common Symptoms Problems*

Installation

Before you begin the actual installation, review the pre-installation guidelines which follow and inspect the package contents.

Pre-Installation Guidelines

- Always use an anti-static wrist strap when handling the unit.
- Do not open the unit cover. Inside parts have hazardous voltages and are extremely sensitive to static. If the unit has been opened, your warranty is void.
- Do not connect equipment in wet conditions and keep away from dusty areas.
- The area must not exceed the temperature and humidity guidelines outlined in *Chapter 2: Hardware Components*.
- Avoid exposing the unit to excessive vibrations.
- Ensure no equipment is put on top of the chassis.

Inspect Package Contents

Before you install the hardware, ensure the following components are included in your shipment:

- Tenor AS
- 1 AC Power adaptor/cord
- 1 DB-9 RS-232 Serial Cable
- RJ-45 Cable
- Product Guide in CD format

If a listed component is not included in your package, contact your customer service representative.

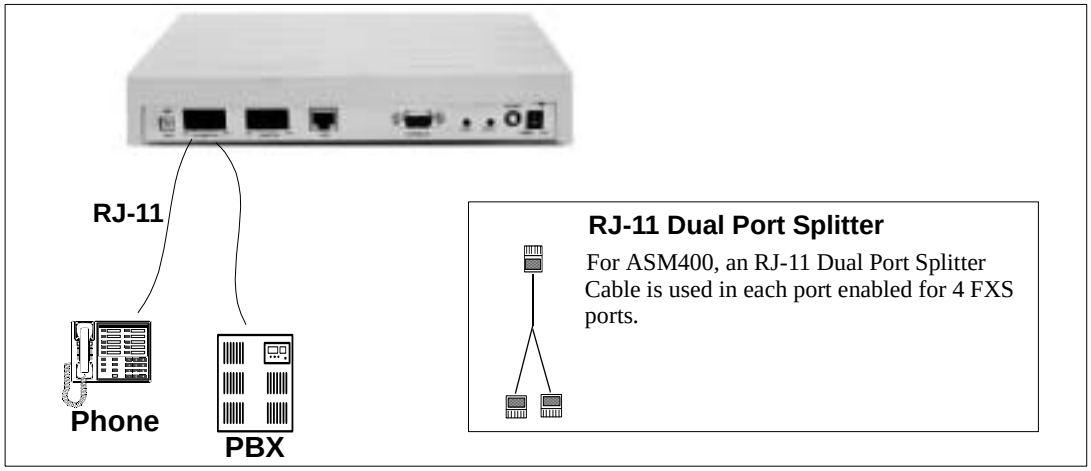
Connect to Phone/FXS Interface

Since there are many different PBX devices, keys systems, fax machines and phones you can connect to the *Tenor AS*, the instructions which follow explain the general procedure for connecting an RJ-11 cable (included in your package) between the Phone/FXS port and an external device. Use the phone/FXS ports for on-premise wiring only.

For the ASM200 and ASG200, you can connect **one** device to each FXS port. For the ASM400 and ASG400, you can connect **two** devices to each FXS port (using an RJ-11 Dual Port Splitter).

See *Chapter 2: Hardware Components* for the RJ-11 cable pinouts you can use to acquire another cable or adaptor that may be required to connect to the specific external device.

Figure 2-1 Connect to Phone/FXO port


RJ-11
Phone
PBX
RJ-11 Dual Port Splitter
For ASM400, an RJ-11 Dual Port Splitter Cable is used in each port enabled for 4 FXS ports.

Connect to Phone/FXS port as follows:

CAUTION: Connect the Phone/FXS ports to a telephone, fax machine, PBX or key system only. Connecting to other devices/networks (i.e., telephone wall jack) will cause damage to the unit.

1. Plug one end of the RJ-11 cable into one of the two ports on the front of the unit labeled *Phone/FXS.* See *Chapter 2: Hardware Components* for cable pinouts.

2. Insert the other end of the RJ-11 cable into the appropriate analog port on the PBX or key system, or into the jack on the back or underside of the phone or fax machine. For the PBX connection, see your PBX documentation port requirements for connection specifics.

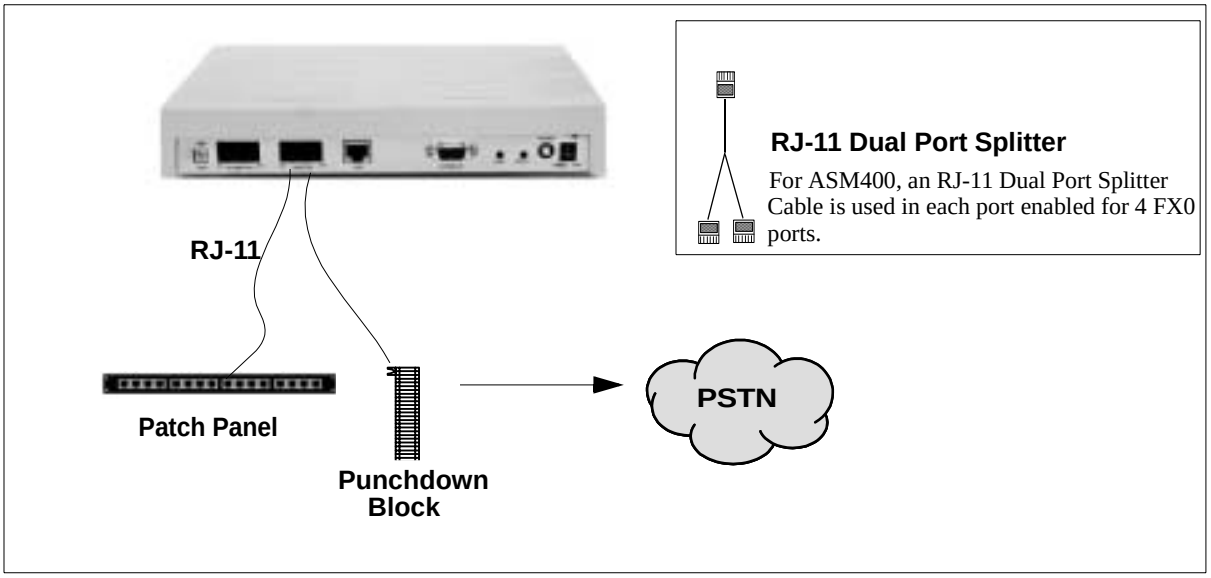
Connect to Line/FXO Interface

To connect to the Line/FXO port, you must first connect the analog phone lines to another piece of equipment that houses your telephone lines running to the PSTN, such as the patch panel, punch down block or wire wrap blocks. If you are unsure of the installation procedures, contact the network administrator or review the documentation you received with the PBX.

For the ASM200 and ASG200, you can connect **one** device to each Line/FXO port. For the ASM400 and ASG400, you can connect **two** devices to each Line/FXO port (using an RJ-11 Dual Port Splitter).

See *Chapter 2: Hardware Components* for the RJ-11 cable pinouts you can use to acquire another cable or adaptor that may be required to connect to the specific external device.

Figure 2-2 Connect to Line/FXO Interface


RJ-11 Dual Port Splitter
For ASM400, an RJ-11 Dual Port Splitter Cable is used in each port enabled for 4 FX0 ports.
RJ-11
Patch Panel
Punchdown Block
PSTN

1. Plug one end of the RJ-11 cable into one of the ports labeled Line/FXO. See *Chapter 2: Hardware Components* for cable pinouts if you are making your own cable.

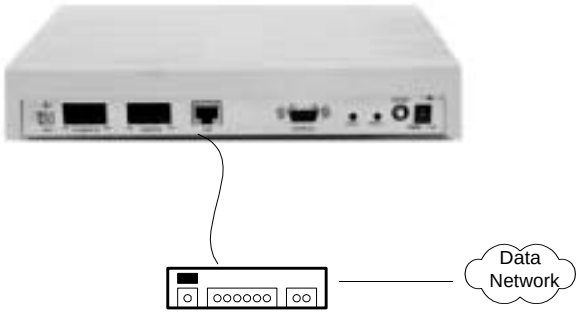
2. Connect the other end of the RJ-45 straight cable to the patch panel (or other external device) which houses your telephone lines.

NOTE: Connecting to the patch panel may require trained telephone personnel.

Connect to Ethernet LAN

You can use these instructions for general connection purposes only. The Ethernet hub/switch manufacturer's documentation should provide specific instructions for connection to another device, such as the *Tenor AS*.

Figure 2-3 Connect to Ethernet Hub/Switch



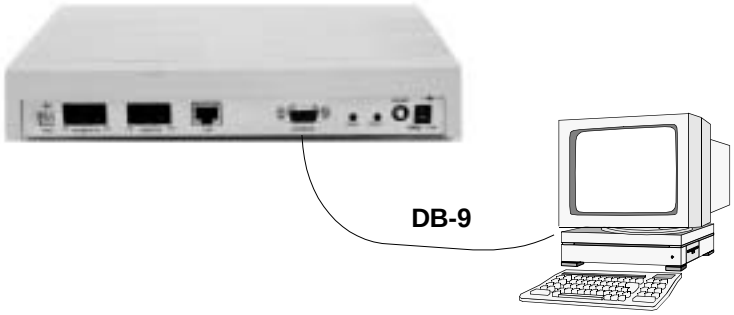
1. Plug one end RJ-45 Ethernet cable into the port labeled LAN.

2. Plug the other end of the cable into one of the Ethernet hub/switch ports. If a custom cable or adapter is required, see *Chapter 2: Hardware Components* for the Ethernet RJ-45 10/100.

Connect to PC Console

You will need to connect the *Tenor AS* to your workstation's serial port via RS-232 connection. (This connection will be used when you assign an IP address to the unit.) For the instructions below, it is assumed you are connecting to a Windows PC.

Figure 2-4 Connect to PC Com Port



1. Insert the male end of the DB-9 cable into the port labeled *Console*. (See *Chapter 2: Hardware Components* for RS-232 connector pinouts.)

2. Insert the female end of the DB-9 cable into your workstation's serial console port (see your PC documentation for more information about this port).

Install Supplemental Ground Safety Cable (if required)

The *Tenor AS* provides a supplemental Earth Ground screw (a #6 screw). This screw provides earth ground to the unit if additional grounding is required. To provide ground via the grounding screw, you will need to connect the grounding screw to a Ground Safety Cable, which can then be connected to an approved safety earth ground.

Connect the Ground Safety Cable as follows:

1. Unscrew the existing screw from the grounding hole.
2. Place the screw through the ring connector at one end of the ground safety cable.
3. Attach the screw securely to the threaded grounding hole.
4. Connect the other end of the ground safety cable to an approved electrically grounded object. Consult with a licensed electrician if you are unclear about this operation.

Figure 2-5 Install Ground Safety Cable

Power up the System

Once you have all cables connected properly, you are ready to turn the system on as follows:

1. Plug in the AC power cord adaptor to an AC outlet.

2. Locate the on/off switch on the back of the unit and click the switch to **On**.

The unit will power up and the LEDs will flash and turn off; the power LED will remain lit. For information about the LEDs, see *Chapter 2: Hardware Components*.

Once the unit is powered up, you are ready to assign an IP address. See the following section *Change IP Address*.

Assign IP address

Before you can configure a *Tenor AS*, you need to assign a valid IP address. When a *Tenor AS* is shipped to a customer, you need to assign a valid IP address for each unit. An IP address is a 32 bit (up to 12 numeric characters) address used to identify each network device in the TCP/IP network. If the unit does not have an IP address, data will not be able to be sent to or from the unit.

Communication between the Tenor and the PC is enabled via RS-232 connection and terminal emulation software. The instructions below assume you are running HyperTerminal (running Windows 95 or later) on your PC. For all other terminal emulation packages, the specific Tenor commands used to assign the IP address will be the same, but the software specific instructions will be different. Consult the applicable documentation for more information.

You can re-configure the IP address using the procedure which follows.

1. Press the Tenor AS's power switch to **On**.

2. Click on *Start> Programs> Accessories> Communications>HyperTerminal> Run.* The *Connection Description* window will be displayed.

3. Enter a connection name (i.e., name for each unit such as *Tenor AS New Jersey*).

4. Click **Ok.**

5. Choose the serial port on your PC from the *Connect Using* drop down list box (i.e., Direct to Com 1). Click **Ok**. The *Com1 Properties* window will be displayed. See Figure 2-6.

Figure 2-6 *Port Settings* Window

6. From the *Bits Per Second* drop down list box, choose *38400*.

7. From the *Data Bits* drop down list box, choose *8*.

8. From the *Parity* drop down list box, choose *None.*

9. From the *Stop bits* drop down list box, choose *1*.

10. From the *Flow control* drop down list box, choose *None*.

11. Click **Ok** and a connection to the Tenor will be established. Information about the unit will scroll on the screen.

12. Enter **login** and **password**. Both are **admin** by default.

13. A message will appear on the screen "Tenor Analog does not have an Ethernet interface configured. Would you like to configure an Ethernet Interface?" (y/n).

14. Type **y**.

15. For *IP Address*, enter the IP address for the Tenor unit.

16. For *Subnet Mask*, enter the subnet mask. This address is used to differentiate the network portion of the IP address from the host portion of the IP address.

17. For *Default Gateway*, choose whether there should be a default gateway (router) which routes packet data outside of your LAN and enter its IP address.

18. A message will appear on the screen "Tenor Analog Ethernet Interface successfully configured." The Tenor will restart using the new Ethernet settings.

Tenor will restart using the new Ethernet settings.

Change IP Address

You are able to change the IP address in which the unit is attached as follows:

NOTE: The instructions below assume you are running Windows 2000 or above.

1. Press the Tenor AS's power switch to **On**.

2. Click on *Start> Programs> Accessories> Communications>HyperTerminal> Run.* The *Connection Description* window will be displayed.

3. Enter a connection name (i.e., name for each unit such as *Tenor AS New Jersey*).

4. Click **Ok.**

5. Choose the serial port on your PC from the *Connect Using* drop down list box (i.e., Direct to Com 1). Click **Ok**. The *Com1 Properties* window will be displayed. See Figure 2-7.

Figure 2-7 *Port Settings* Window

6. From the *Bits Per Second* drop down list box, choose *38400*.
7. From the *Data Bits* drop down list box, choose *8*.
8. From the *Parity* drop down list box, choose *None.*
9. From the *Stop bits* drop down list box, choose *1*.
10. From the *Flow control* drop down list box, choose *None*.
11. Press the *Tenor AS* power switch to **On**. After the bootup sequence, the login prompt will appear.
12. Enter a login name. The default login name is *admin.*
13. Enter a password. The default password is *admin*. (Once you are up and running, changing the password is a good idea for security purposes). Step through each of the following parameters and enter the correct values for your installation: IP address, Subnet Mask and Default Gateway.
14. At the **Quintum** prompt, type **ei** to reach the Ethernet prompt and then type **config** to change to the Configuration mode.
15. To set the IP address, type **set ipa** followed by the IP address.
16. To set the Subnet Mask, type **set subnetmask**, followed by the subnet mask.
17. Type **siprd** to change to the Static IP Route Directory.
18. To set the Default Gateway IP, type **change 1 g** followed by the IP address for the default gateway IP.
19. Type **submit**.
20. Type **maint** to reach the maintenance mode and then **mc**. Type **reset**. A confirmation message will ask if you want to reset the unit. Type **yes** to reset the unit. The reboot enables the Tenor to incorporate the new settings.

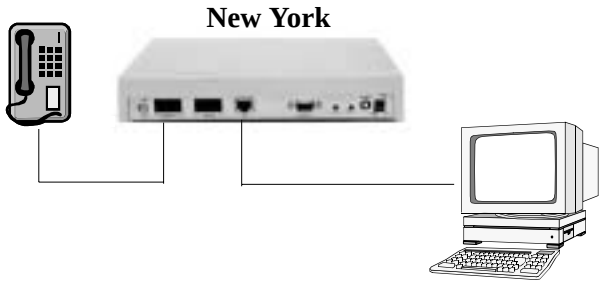
Getting Started with Configuration/Making the First Call

This section includes basic information for making the first call using VoIP and a Tenor AS; for an example, this call will enable you to dial Quintum Technology's test unit and hear a recorded message. Once you have accomplished that, you can modify the configuration to meet your own specific needs.

For future calls, if you require detailed configuration information, see the *Tenor Configuration Manager/Tenor Monitor User's Guide* and *Command Line Interface (CLI)* guide for detailed configuration information (both documents are located on the CDR ROM you received with the unit or you can download the latest documentation from *www.quintum.com).*

If you are using a cable modem, or a DSL modem and a firewall, specific instructions are included in this section, When configuring a firewall, set up a DMZ (this makes the firewall act as a switch so that all incoming IP traffic for the firewall's WAN IP will be routed directly to the Tenor AS). If you are using a cable modem with NO firewall, specific configuration options are also included.

Figure 2-8 Making the First Call



These instructions assume the unit is taken right from the box and is in the default state from the factory, and basic connections are made. See below for a list of prerequisites that must be met before making a call from the Tenor.

- An analog phone connected to the port labeled FXS.
- A connection between the Tenor's port labeled *Console* and your workstation's serial console port. See *Installation.*
- Tenor *Configuration Manager* software is loaded on your workstation.
- Through HyperTerminal, you have configured an Ethernet Interface by assigning an IP Address, Subnet Mask, and Default Gateway (see *Assign IP address* for more information).

Execute the first call as follows:

NOTE: If the call does not connect or you encounter a problem, see *Common Symptoms/Problems* or *Chapter 5: Advanced Topic: Diagnostics/Maintenance* for possible solutions.

1. Start the *Configuration Manager.*

2. At the prompt, enter the **IP address** for your unit and enter the default login and password (**admin/admin**).

3. At the *Phone(FXS)/Line(FXO)> AnalogInterface-Line* screen, enable all four lines (or two lines, depending upon the unit type) under **FXO Channel Assignment.** Click on **Confirm/OK.**

4. At the *System-Wide Configuration> Dial Plan* screen, enter an **Area Code** (default is 732) and check **Use 10 Digit Local Dial.** Click on **Confirm/OK.**

   Allows the unit to use the 732 area code for its call without having to dial a "1".

5. At the *System-Wide Configuration> Time Server screen*, set the **Primary Time Server IP Address** to a standard Time Server IP Address. This is not a required entry for the first call, but it is a good idea to set it at this point.

   Set the **Secondary Time server IP Address** to a standard Time Server IP Address.

   Set the **UTC Offset** to -5 hours.

   Click on **Confirm/OK.**

6. At the *VoIP Configuration> Gatekeeper/Border Element* screen, set the **Primary Border Element IP Address** (i.e., 208.226.140.40 for Quintum's test unit). Click on **Confirm/OK.**

7. At the *VoIP Configuration> H323 Signaling Group* screen, set the **Primary Gatekeeper IP** (i.e., 208.226.140.40 for Quintum's test unit) Click on **Confirm/OK.**

   Allows the unit to get the information it needs to route your phone call to a specific IP address.

8. At the *System-Wide Configuration> Circuit Configuration> Line Routing Configuration> Hunt LDN Directories> Hunt LDN Directory-pub 1* screen, enter a **pattern**. (i.e., 10 digit phone number that will be assigned to the phone on your end). Click on **Confirm/OK.**

   This provides the Gatekeeper with a phone number designation for your unit.

For use with a Cable Modem or DSL Modem and a Firewall only

9. At the *Ethernet Configuration> Ethernet Interfaces> Ethernet Interface-1* screen and enter the **External NAT IP** (this is the IP that the service provider assigns to the firewall WAN port). Click **Confirm/OK**. This was tested with a Linksys™ firewall. Other types may work as well and should be configured similarly.

For use with a Cable Modem and No Firewall only

10. At the *Ethernet Configuration> Ethernet Interfaces> Ethernet Interface-1* screen, check the **Enable DHCP** box. Click **Confirm/OK**.

    This provides a way for your service provider to assign an IP to your unit which can then be reached by the Gatekeeper on the public internet.

11. Submit all changes (through *File>Submit Changes*).

12. Pick up the phone and hear dial tone. Dial 7324609000. The call should route to Quintum's test unit and you should hear a recorded message.

    When you dial 7324609000 (ten-digits), your unit (the Gateway) consults the Gatekeeper's table of phone number to IP translations (the Gatekeeper is an application or function

inside the test Gateway at 208.226.140.40), and sends the appropriate data to the other endpoint (in this case, the Quintum test unit). The Gateway is merely another function or application of 208.226.140.40. The Gateway functions allow the incoming call to be connected to Quintum's PBX.

Load Software Upgrade

To upgrade the software, download the upgrade from the CD ROM you received with the unit, or download the latest software/documentation from *www.quintum.com*.

Common Symptoms/Problems

Before you begin troubleshooting a potential malfunction, it is a good idea to check your basic hardware connections. See below.

- Ensure power cord is firmly installed in the back panel's power jack and the other end is plugged into the AC power source.
- Ensure the unit's power switch is in the On position.
- Verify that all RJ-45, RJ-11 and DB-9 cables fit snugly in each back panel jack. Faulty connections may cause a number of network interfacing or connection issues.

If you suspect the problem to be on the network end, contact your Central Office to verify proper operation.

Below is a list of common symptoms and problems you may encounter. Use this list as a guideline; if your problem is not listed, use the diagnostic procedure explained in the beginning of this chapter.

Table 2-1 Common Symptoms/Problems

Common Symptom/Problem	Description/Solution
Unit will not turn on.	Check AC power source.
Communication between *Tenor AS* and the FXO or FXS cannot be established.	There are several reasons why communication may not be successful. A few of the most common are listed below. Verify correct RJ-11, RJ-45 and DB-9 cables are installed in the Phone/FXS and Line/FXO ports. Ensure the unit is on. Network issues may cause a number of problems. Contact the Central Office to perform test procedures.
Communication with *Command Line Interface (CLI)* cannot be established using Telnet.	The IP address of the *Tenor AS* unit may be incorrect. Check Ethernet Cable. Verify the IP address of *Tenor AS*. Check the Default Gateway Subnet Mask. Check Ethernet connection via RS-232 connection. See *Chapter 3: Installation.* Verify network connectivity using *ping* from another network host.

Common Symptom/Problem	Description/Solution
Communication with Ethernet Hub, or switch cannot be established.	Verify RJ-45 cable is firmly installed in the Ethernet port. Check MDI/MDIX configuration. Check duplex setting on the switch in which they were connected and the speed of 10MB or 100 MB.
Communication between the computer's COM port and *Tenor AS* serial port cannot be established.	Verify DB-9 cable is firmly placed in the unit's console port and your PC's serial port. Verify Terminal port settings at 38400 BPS 8N1 No Flow Control.
Tenor AS cannot receive or transmit calls.	Ensure FXO/FXS ports are working correctly. Generate alarm list for more information. Contact Central Office for interface issues.

Chapter 4: Advanced Topic: View Call Detail Records

This chapter tells you how to display and understand the Call Detail Recording (CDR) feature, which enables you to view call information. Examples are included later in this chapter.

Specifically, the following topics are included.

- *CDR Description*
- *Establish Connection*
- *CDR Output*

What is a CDR?

A Call Detail Record (CDR) is a string of data which contains call information such as call date and time, call length, calling party, and called party. Through the Call Detail Recording (CDR) feature, the *Tenor AS* unit is able to generate a CDR at the completion of each call. CDRs are collected from multiple *Tenor AS* units simultaneously and continuously.

A CDR file can be created each day to collect CDRs from each *Tenor AS* that connects to the server. From this information you can capture billing type data which can be used by separate software components to create billing reports, view call records, and generate daily/weekly/monthly statistics reports.

The last 9600 CDRs generated are stored by the *Tenor AS* unit in a circular buffer (this means that any number of CDRs over 9600 will overwrite the existing CDRs). We advise that you set up a PC or workstation to act as a CDR server responsible for receiving the CDRs as they are generated (up to two ports can be set up to collect CDRs from the *Tenor AS* unit). The server will be responsible for capturing CDRs via TCP/IP, processing/storing them in permanent memory, and producing billing records. Any CDRs not collected from the *Tenor AS* unit will be lost if the unit is powered down.

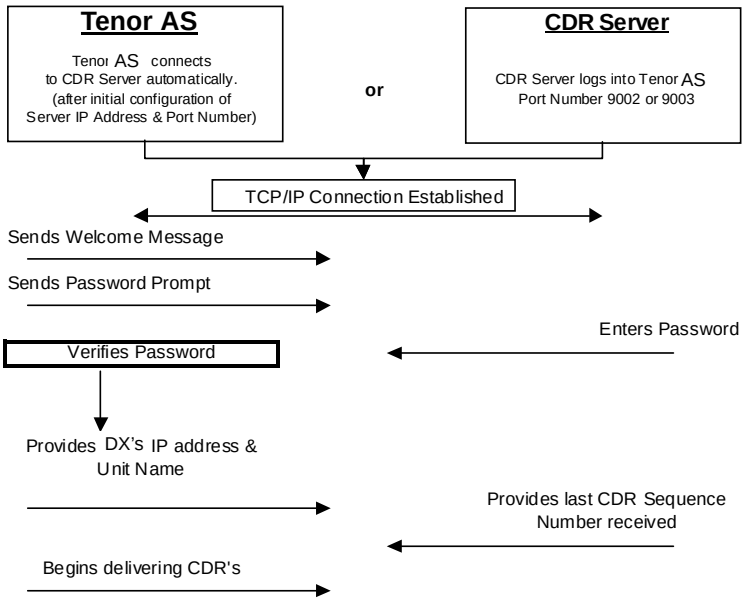
The CDR software and Billing software mentioned is 3rd party software, and is not supported by Quintum.

There are two ways to view Call Detail Records (CDRs) for the *Tenor AS* unit: through the *Command Line Interface (CLI)* or through *Tenor Monitor.* The information for accessing CDRs via CLI is detailed in this chapter: see the *Tenor Configuration Manager/Tenor Monitor* User's Guide for information about viewing alarms via *Tenor Monitor.*

Establish connection between *Tenor AS* and CDR Server

In order to capture CDRs, a connection between the *Tenor AS* unit and the CDR server must be established. A *Tenor AS* can be configured to connect up to two CDR servers via port 9002 and 9003. Based on configuration, the *Tenor AS* unit can either establish a TCP/IP session with one or all of these CDR servers. A flow diagram (Figure 4-1) illustrates the general transfer of information.

Figure 4-1 Flow of CDR Information


Tenor AS
Tenor AS connects to CDR Server automatically. (after initial configuration of Server IP Address & Port Number)
or
CDR Server
CDR Server logs into Tenor AS Port Number 9002 or 9003
TCP/IP Connection Established
Sends Welcome Message
Sends Password Prompt
Enters Password
Verifies Password
Provides DX's IP address & Unit Name
Provides last CDR Sequence Number received
Begins delivering CDR's

Before attempting to collect CDRs, you should configure the desired information. You can assign a CDR server IP address, CDR server port number, CDR server password, and CDR format information using the following CLI commands: *cdrserverip*, *cdserverport*, *cdrpassword*, and *cdrformat*.

- **CDRServerIPAddr:** IP address of the CDR server. (Used when the *Tenor AS* unit established connection with the CDR server.)
- **CDRServerPort:** The application port numbers used by the CDRServer(s). (Used when the *Tenor AS* establishes connection with the CDR server.)
- **CDRPassWord:** Password to be used by the CDR server(s).
- **CDRFormat:** This configuration parameter command is used to choose which of the possible Call Data record output formats you would like to send to your CDR server. See *CDR Output* for information about available formats.

Configure *Tenor AS* for connection to CDR Server

NOTE: The CDR Server software is a Windows-based *.exe* file available on the CD you received with your system; this software is not supported by Quintum.

The instructions below are performed via *Command Line Interface (CLI)*. See the *Command Line Interface (CLI)* user guide for specific information.

1. Through CLI, access the *Configuration* prompt.

2. Access the *config-CDRServer-1#* prompt (the number will change according to the desired server).

3. Type *set CDRServerIPaddr* followed by the IP address of the CDR server and press **Enter**.

4. Type *set CDRServerPort* followed by the desired port number and press **Enter.** The default port is 9002.

5. Type *set CDRPassWord* followed by the desired password and press **Enter**. The password is an alphanumeric string.

6. Type *set CDRFormat* followed by the desired format (0, 1, 3, 4, 100, 101, 103 or 104) for displaying CDRs. See previous section for definitions.

Once you configure this information, you will be able to capture CDR reports through the CDR server and the *Tenor AS* unit, and it will be able to establish a TCP/IP session with the server on its own.

Setup CDR Server and assign password

Before the CDR server can collect CDRs, you must install the *cdrserver.cfg* file *as follows:*

NOTE: The software is a Windows-based *.exe* file available on the CD you received with your system; this software is not supported by Quintum.

1. Create a directory in which to install the *cdrserver.cfg* file, such as c:\cdr.

NOTE: You can obtain the *cdrserver.cfg* file via Quintum Training class or through our web site at *www.quintum.com.*

2. Copy the *cdrserver.cfg* file and install it into the directory you created in step 1.

3. Copy the *cdrsrv.exe* file and install it into the directory your created in step 1.

4. Double-click on the *cdrserver.exe* file. The CDR files will be generated and saved to the directory in which you are working. File names are listed as the data/gateway from which the file was created.

Change CDR Password

Change the CDRserver password, if desired, as follows:

1. From the directory in which you are working, right click on the *cdrserver.cfg* file. At the *Open with* option, choose Notebook.

2. Scroll down to the line stating *cdr_password.* Next to that line, enter the password. Valid entry: up to 30 characters.

Tenor AS Establishes Connection with CDR Server

To capture CDR reports via CDR server (*i.e.,* a PC or workstation you use to capture CDR data) you must first configure the IP address and port number of the CDR server in the *Tenor AS* unit. Once these are configured, the *Tenor AS* unit will be able to establish a TCP/IP session with the server on its own.

CDR Server Establishes Connection with *Tenor AS*

If an IP address/port number is not configured, the CDR server has to initiate the session. The *Tenor AS* unit uses TCP port numbers 9002 and 9003 on its side for the CDR sessions. The CDRServerport must still be configured to either 9002 or 9003 (see the *Command Line Interface* user guide for specific information).

Whether the *Tenor AS* unit establishes the connection with the CDR server or the CDR server attempts to establish the connection, there is a limit of 5 attempts to enter the correct password before the TCP session is terminated (to configure a password, see the CLI command *cdrpassword* in the previous section).

After the CDR server successfully logs into the CDR port of the *Tenor AS* unit, the CDR server will be provided with the IP address and unit name of the *Tenor AS*. The CDR server will then supply the sequence number of the last CDR that it has received from the *Tenor AS* unit. If the last CDR number is unknown, the server should send 0 for the sequence number. After this exchange, the *Tenor AS* will start delivering new CDRs to the server.

CDR Output

The following is an example of a CDR output. Each field in a CDR string is separated by a comma (any blank fields are designated by a comma). See below for field definitions.

There are four CDR format types:

- 0 (Standard CDR output)
- 1 (Extended CDR output)
- 3 (Extended Tenor AS output and CDR output)
- 4 (Extended Tenor AS output and CDR output)
- 100 (Standard CDR output same as selection 0 output plus session ID)
- 101 (Extended CDR output same as selection 1 output plus session ID)
- 103 (Standard CDR output same as selection 3 output plus session ID)
- 104 (Extended CDR output same as selection 4 output plus session ID)

Sample Record for Standard and Extended CDR Format 0, 1, 100, 101

Record 1 Sample: (includes fields for both Standard and Extended Formats)

1,17325551212,15,20000207062812,21060207062815,
2000020706283030,16,208.226.140.57,192.168.10.64,4,1,1,1,1,1,1,,0,0,1415551000,12345678901234,9876543210,0123456789

Record 1 Field Definitions – Standard Formats 0 and 100

1 **(Call ID),** 17325551212 **(Called Number)**, 15 **(Duration),** 20000207062812 (**Call Initiation Time**), 20000207062815 **(Call Connected Time)**, 2000020706283030 **(Call Disconnected Time)**, 16 **(Cause Code)**, 208.226.140.57 **(Local IP Address)**, 192.168.10.64 **(Remote IP Address)**, 4 **(Origination Trunk ID)**, 1 **(Call Type)**, 1 **(Call Number Type)**, 1 **(Incoming Line)**,1 **(Incoming Channel)**, 1 **(Outgoing Line)**, 1 **(Outgoing Channel)**, blank **AutoSwitch Time**, blank (**AutoSwitch Duration),** 0 **(Bad IP Quality Events),** 0 **(AutoSwitch Flag)**

Record 1 Field Definitions – Extended Formats 1 and 101

The extended format includes all fields used in the standard format plus the following fields:

1415551000 **(Calling Party Number),** 12345678901234 **(PIN Code),** 9876543210 **(Remote Call ID #)**, 0123456789 **(Local Call ID #)**

Definitions for each field appear below.

Call ID: Sequence number. This is a unique number assigned to identify an individual call (i.e, 1, 2, 3,…). The sequence number starts from 1 and wraps around at 4,294,967,295. When a *Tenor AS* unit resets, the sequence number starts from 1 again. If the system has a problem and loses connectivity,

the CDR server can send the *Tenor AS* unit the last Call ID that is received. The *Tenor AS* unit will reply with all records that contain a Call ID which is greater than the one last received.

Called #: The number called. This will be in international format except for a pass-through call going from PBX to PSTN or a call going from PSTN to PBX.

Duration: Call duration. This value is in seconds, the value will be 0 if never connected.

Call Initiation Time: The date and time the call initiated. The time will be the local time configured on the *Tenor AS* unit. The entry will be in the following format: yyyymmddhhmmss where yyyy (4 digits for year), mm (2 digits for month), dd (2 digits for day), hh (2 digits for hour), mm (2 digits for minutes), ss (2 digits for seconds).

Call Connected Time: The date and time the call was actually connected. The time will be the local time configured on the *Tenor AS* unit. The entry will be in the following format: yyyymmddhhmmss where yyyy (4 digits for year), mm (2 digits for month), dd (2 digits for day), hh (2 digits for hour), mm (2 digits for minutes), ss (2 digits for seconds). This field will blank if the call never connected.

Call Disconnected Time: The date and time the call disconnected. The time will be the local time configured on the *Tenor AS* unit. The entry will be in the following format: yyyymmddhhmmss where yyyy (4 digits for year), mm (2 digits for month), dd (2 digits for day), hh (2 digits for hour), mm (2 digits for minutes), ss (2 digits for seconds).

Cause Code: The Q.931 cause value assigned if the call is not connected. Possible common entries are listed below. This field will be blank if the call was connected.

Cause Code	Definitions
16	**Normal Call Clearing.** The cause indicates that the call is being cleared because one of the users has requested that the call be cleared.
17	**User Busy.** The called system acknowledges the connection request but is unable to accept the call because all B channels are in use.
18	**No User Responding.** This cause is used when a user does not respond to a call establishment message with either an alerting or connect indication within the prescribed period of time allocated (in Q.931 by the expiry of either timer T303 or T310).
28	**Invalid Number Format (Address Incomplete).** The cause indicates that the called user cannot be reached because the called party number is not a valid format or is not complete.
31	**Normal, Unspecified.** This cause is used to report a normal event only when no other cause in the normal class applies.
34	**No Circuit/Channel Available.** The connection cannot be established because no appropriate channel is available to take the call.

Cause Code	Definitions
47	**Resource Unavailable, Unspecified.** This cause is used to report a resource unavailable event only when no other cause applies.

Local IP Address: The IP address for the *Tenor AS* unit originating the CDR. The entry will be in the following format: xxx.xxx.xxx.xxx.

Remote IP Address: IP address for the remote destination *Tenor AS*. This will be generated only if the call is VoIP; if the call is circuit based, this field will be blank. The entry will be in the following format: xxx.xxx.xxx.xxx.

Origination Trunk ID: Identifies the origination trunk ID of the line that initiated the call. This field will be blank if no trunk ID is configured. The trunk ID is often configured with an account code.

Call Type: The type of call. Valid entry: 1 = voice, 2 = fax, 3 = modem, 4 = data.

Call Number Type: The called numbering plan used for the call per Q.931. Possible common entries are as follows: 1 = Public/E.164, 9 = Private.

Incoming Line: If the call is incoming, this field identifies which line the call came in on. Valid entry: 1 = PBX, 2 = PSTN. This field will be empty if the call is an incoming VoIP call.

Incoming Channel: If the call is incoming, this field identifies which channel the call came in on. Valid entry: 1-2 or 1-4. This field will be empty if the call is an incoming VoIP call.

Outgoing Line: If the call is outgoing, this field identifies which line the call is going out on. Valid entry: 1 = PBX, 2 = PSTN. This field will be empty if the call is an outgoing VoIP call.

Outgoing Channel: If the call is outgoing, this field identifies which channel the call went out on. Valid entry: 1-2 or 1-4. This field will be empty if the call is an outgoing VoIP call.

Autoswitch Time: This is the date and time the autoswitched occurred (when the call is switched from VoIP to circuit). The entry will be in the following format: yyyymmddhhmmss where yyyy (4 digits for year), mm (2 digits for month), dd (2 digits for day), hh (2 digits for hour), mm (2 digits for minutes), ss (2 digits for seconds). If an autoswitch did not occur, this field will be blank.

AutoSwitch Duration: The number of seconds that the autoswitch call was active. Valid entry: xx.

Bad IP Quality Events: The number of bad quality events that occur during a VoIP call. This number determines the overall quality of the call.

Autoswitch Flag. The terminating side of the autoswitch call initially terminates to the internal autoswitch agent before the call is actually autoswitched. This termination generates an extra CDR in addition to the original call that is autoswitched. This field identifies a call that is terminated to the autoswitch agent, 0 = normal call, 1 = termination to the autoswitch agent.

The following are CDR fields used in the extended format (1 or 101) only.

Calling Party Number. The number called from. The format will be delivered in whatever format the PSTN or PBX delivers to the *Tenor AS*.

PIN Code. PIN code entered. 14 digits maximum. This field will be blank if a PIN code is not configured.

Remote Call ID #. Unique identification number, generated by the remote-side *Tenor AS*, for call record matching purposes. Only generated for IP calls. For a given IP call, the Local Call ID on one Tenor should match the Remote Call ID of the other.

Local Call ID #. Unique identification number, generated by the local-side *Tenor AS*, for call record matching purposes. Generated for all IP calls. For a given IP call, the Local Call ID on one Tenor should match the Remote Call ID of the other.

Sample Record for Extended Tenor AS CDR Format 3, 4, 103, 104:

Record 1 Sample: (includes fields for formats 3 and 103)

1,17325551212,15,20000207062812,21060207062815,2000020706283030,16,208.226.140.57,192.168.10.64,4,1,1,1,2,0,1,1,2,0,1,1,,0,0,1415551000,12345678901234,9876543210,0123456789,12138765432

Record 1 Field Definitions - Tenor AS Extended Formats (3 and 103)

1 **(Call ID),** 17325551212 **(Called Number)**, 15 **(Duration),** 20000207062812 (**Call Initiation Time**), 20000207062815 **(Call Connected Time)**, 2000020706283030 **(Call Disconnected Time)**, 16 **(Cause Code)**, 208.226.140.57 **(Local IP Address)**, 192.168.10.64 **(Remote IP Address)**, 4 **(Origination Trunk ID)**, 1 **(Call Type)**, 1 **(Call Number Type)**, 2 **(Incoming Slot),** 0 **(Incoming Device),** 1 **(Incoming Analog Interface)** 1 **(Incoming Channel),** 2 **(Outgoing Slot),** 0 **(Outgoing Device),** 1 **Outgoing Analog Interface),** 1 **(Outgoing Channel),** blank **(AutoSwitch Time),** blank **(AutoSwitch Duration),** 0 **(Bad IP Quality Events),** 0 **(AutoSwitch Flag),** 1415551000 **(Calling Party Number),** 12345678901234 **(PIN Code),** 0123456789 **(Local Call ID #),** 9876543210 **(Remote Call ID #)**

The 4 and 104 extended format includes all fields used in the 3 and 103 extended format plus the following field:

12138765432 **(Incoming/Outgoing IP DN).**

Definitions for each field appears below.

Call ID: Sequence number. This is a unique number assigned to identify an individual call (i.e, 1, 2, 3,…). The sequence number starts from 1 and wraps around at 4,294,967,295. When a Tenor unit resets, the sequence number starts from 1 again. If the system has a problem and loses connectivity, the CDR server can send the *Tenor AS* unit the last Call ID that it received. The *Tenor AS* unit will reply with all records that contain a Call ID which is greater than the one last received.

Called #: The number called. This will be in international format except for a pass-through call going from the PBX to the PSTN or a call going from the PSTN to the PBX.

Duration: Call duration. This value is in seconds, the value will be 0 if never connected.

Call Initiation Time: The date and time the call initiated. The time will be the local time configured on the *Tenor AS* unit. The entry will be in the following format: yyyymmddhhmmss where yyyy (4 digits for year), mm (2 digits for month), dd (2 digits for day), hh (2 digits for hour), mm (2 digits for minutes), ss (2 digits for seconds).

Call Connected Time: The date and time the call was actually connected. The time will be the local time configured on the *Tenor AS* unit. The entry will be in the following format: yyyymmddhhmmss where yyyy (4 digits for year), mm (2 digits for month), dd (2 digits for day), hh (2 digits for hour), mm (2 digits for minutes), ss (2 digits for seconds). This field will be blank if the call never connected.

Call Disconnected Time: The date and time the call disconnected. The time will be the local time configured on the *Tenor AS* unit. The entry will be in the following format: yyyymmddhhmmss where yyyy (4 digits for year), mm (2 digits for month), dd (2 digits for day), hh (2 digits for hour), mm (2 digits for minutes), ss (2 digits for seconds).

Disconnect Cause Code: The Q.931 cause value assigned if the call is not connected. Possible common entries are listed below. This field will be blank if the call was connected.

Cause Code	Definitions
16	**Normal Call Clearing.** The cause indicates that the call is being cleared because one of the users has requested that the call be cleared.
17	**User Busy.** The called system acknowledges the connection request but is unable to accept the call because all B channels are in use.
18	**No User Responding.** This code is used when a user does not respond to a call establishment message with either an alerting or connect indication within the prescribed period of time allocated (in Q.931 by the expiry of either timer T303 or T310).
28	**Invalid Number Format (Address Incomplete).** The cause indicates that the called user cannot be reached because the called party number is not a valid format or is not complete.
31	**Normal, Unspecified.** This code is used to report a normal event only when no other cause in the normal class applies.
34	**No Circuit/Channel Available.** The connection cannot be established because no appropriate channel is available to take the call.
47	**Resource Unavailable, Unspecified.** This code is used to report a resource unavailable event only when no other code applies.

Local IP Address: The IP address for the *Tenor AS* unit originating the CDR. The entry will be in the following format: xxx.xxx.xxx.xxx.

Remote IP Address: IP address for the remote destination *Tenor AS*. This will be generated only if the call is VoIP; if the call is circuit-based, this field will be blank.

Origination Trunk ID: Identifies the origination trunk ID of the line that initiated the call. This field will be blank if no trunk ID is configured. The trunk ID is often configured with an account code.

Call Type: The type of call. Valid entry: 1 = voice, 2 = fax, 3 = modem, 4 = data.

Call Number Type: The called numbering plan used for the call per Q.931. Possible common entries are as follows: 1 = Public/E.164, 9 = Private.

Incoming Slot: The slot number to which a call enters. This entry is fixed at 2.

Incoming Device: If the call is incoming, this field identifies which Analog Interface the call came in on. This entry is fixed at 0.

Incoming Analog Interface: If the call is incoming, this field identifies which device interface the call came in on. Valid entry: 1 = PBX, 2 = PSTN. This field will be empty if the call is an incoming VoIP call.

Incoming Channel: If the call is incoming, this field identifies which channel the call came in on. Valid entry: 1-2 or 1-4. This field will be empty if the call is an incoming VoIP call.

Outgoing Slot: If the call is outgoing, this field identifies the slot the call is going out on. This entry is fixed at 2.

Outgoing Device. If the call is outgoing, this field identifies the Device the call is going out on. This field is fixed at 0.

Outgoing Analog Interface. If the call is outgoing, this field identifies which Analog Interface the call is going out on. Valid entry: 1 = PBX, 2 = PSTN. This field will be empty if the call is an outgoing VoIP call.

Outgoing Channel. If the call is outgoing, this field identifies which channel the call went out on. Valid entry: 1-2 or 1-4. This field will be empty if the call is an outgoing VoIP call.

Autoswitch Time: This is the date and time the autoswitched occurred (when the call is switched from VoIP to circuit). The entry will be in the following format: yyyymmddhhmmss where yyyy (4 digits for year), mm (2 digits for month), dd (2 digits for day), hh (2 digits for hour), mm (2 digits for minutes), ss (2 digits for seconds). If an autoswitch did not occur, this field will be blank.

AutoSwitch Duration: The number of seconds that the autoswitch call was active. Valid entry: xx.

Bad IP Quality Events: The number of bad quality events that occur during a VoIP call. This number determines the overall quality of the call.

Autoswitch Flag. The terminating side of the autoswitch call initially terminates to the internal autoswitch agent before the call is actually autoswitched. This termination generates an extra CDR in addition to the original call that is autoswitched. This field identifies a call that is terminated to the autoswitch agent, 0 = normal call, 1 = termination to the autoswitch agent.

Calling Party Number. The number called from. The format will be delivered in whatever format the PSTN or PBX delivers to the *Tenor AS*.

PIN Code. PIN code entered. 14 digits maximum. This field will be blank if a PIN code is not configured.

Local Call ID #. Unique identification number, generated by the local-side *Tenor AS*, for call record matching purposes. Generated only for IP calls.

Remote Call ID #. Unique identification number, generated by the remote-side *Tenor AS*, for call record matching purposes. Only generated for IP calls.

The following is a CDR field used in the extended format 4 and 104:

Incoming/Outgoing IP DN. If this is an incoming IP call, the number displayed will be the number as received from the other endpoint. If this number is an Outgoing IP call, the number displayed will be the DN as it was sent out over IP (Outgoing number plus prepended digits).

Chapter 5: Advanced Topic: Diagnostics/ Maintenance

This chapter explains the advanced topics for diagnosing problems with the unit, viewing system alarms, and performing maintenance procedures, such as upgrading the software.

- ❑ *Monitor LEDs*
- ❑ *Monitor Alarms*
- ❑ *Verify Unit Provisioning*
- ❑ *Perform Maintenance*
- ❑ *Find Additional Help*

Monitor LEDs

LEDs monitor the health of the system; they are the first signal that the unit is not working properly or that an internal or external error has occurred. LEDs appear on the front of the unit. See *Chapter 2: Hardware Components* to ensure the correct lighting of each LED. If the LEDs are not lighting at all, check the AC power source to ensure power is being supplied to the unit.

Monitor Alarms

Alarms are brief text messages that appear on your workstation when the *Tenor AS* unit encounters a problem, such as a failed interface, disconnected call, etc. Alarms help you identify where a specific problem is occurring with the *Tenor AS* unit.

There are two ways to view alarms for the *Tenor AS* unit: through the *Command Line Interface (CLI)* or through *Tenor Monitor.* The instructions in this section are for obtaining alarms through the CLI; see the *Tenor Configuration Manager/Tenor Monitor* User's Guide for information about obtaining alarms through the *Tenor Monitor.*

How to Read Alarms

The *Alarm Manager* reports alarms according to criteria such as the alarm's severity level, line number the alarm occurred on, channel number, etc. There are two alarm types displayed: Active Alarms and Alarm History. An Active Alarm list displays all the alarms still active on the system; these alarms have not been cleared or deleted. An Alarm History is a list of the last 100 alarms stored in the system since the last time you performed a delete operation.

Definitions for generated alarm fields appear in Table 5-1.

Table 5-1 Alarm Fields and Definitions

Field	Definition	Valid Entry
IP #	The unit's IP address (32 bit address).	Example:192.168.1.34.
Sequence #	Internal number used to identify alarms.	01, 02, 03, etc.

Field	Definition	Valid Entry
Type (displays only if you generate an Alarm History)	The type of alarm generated.	ALR = Alarm. This indicates an active alarm. CLR= Clear. This indicates an alarm that has been cleared from the system. RPT= Report. This indicates that the alarm has been generated for a report. This entry is for internal use only; if you see an alarm that is causing problems, contact customer service.
Severity	Level or alarm severity.	1 = Critical (complete system is affected). 2 = Major (major problem is detected). 3 = Minor (minor problem is detected). 4 = Info (Information about a minor problem).
Description	A text description of the alarm; see Table 5-2 for detailed description.	Varies.
Slot #	Defines which slot the alarm occurred on.	Slot 1 or 2. Slot 1 refers to the system controller functions; slot 2 refers to DSP functions.
Device #	Defines which device the alarm occurred on.	Always device 0.
Analog Interface	Defines which interface (line) alarm occurred on.	1 = PBX 2 = PSTN
Channel #	Specifies which channel the alarm occurred on.	Channels 1-2 or 1-4, depending on product type.
Date/Time	Date/time the event occurred on.	Day of week: name of day. Month: Jan, Feb, March, etc. Day of month: 1 or 2 digits. Time: 6 digits (hour minutes seconds based on a 24-hour clock). Year: 4 digits.

Valid Alarms

The following is a list of all alarm descriptions (text that appears in the Alarm Description field) for all possible alarms the system can generate. In the generated alarm list, the alarm description appears as part of the Description field.

Table 5-2 List of Valid Alarms

Severity (appears as part of severity field)	Alarm Description (text appears in desc field)	Definition
Critical	Remote Alarm indication (Yellow Alarm)	*Tenor AS* is receiving a yellow alarm signal from the network.
Critical	Loss of signal	A loss of signal (32 consecutive zeros) at least once during a one second period.
Critical	Ethernet Disconnected	Ethernet cable has been disconnected from the unit, or Ethernet connectivity has been lost. No new VoIP calls will be made and existing PSTN calls will be switched to the PSTN.
Critical	Call Handler not registered with Gatekeeper	The Call Handler process cannot be registered with the Gatekeeper.
Critical	Critical Software Error	A software error has occurred that affects the operability of the complete system.
Critical	*Tenor AS* Chassis reset	The chassis has reset.
Critical	Configuration Data Missing	Configuration via CLI is missing. Check the configuration data and add the necessary information.
Critical	IVR Configuration Missing	Appears if an attempt to make an IVR call has been made when a valid IP address is not configured. Occurs if an IVR call has been passed through accidently, without a real intention to use IVR for subsequent calls, while both of the servers were disabled. In order to clear the alarm, a user will have to change one of the IP addresses to some value, and then disable it again.
Critical	RADIUS Configuration Missing	Appears when a RADIUS request is made and one or more required configuration parameters are missing. This alarm is cleared when the required RADIUS parameters are configured via CLI.

Severity (appears as part of severity field)	Alarm Description (text appears in desc field)	Definition
Critical	RADIUS Server Not Responding	Appears when none of the configured RADIUS servers respond. This alarm is cleared when any of the RADIUS servers responds or the RADIUS server is disabled via CLI.
Major	Major Software Error	A software error has occurred that affects system signaling, interfaces, or other major operation.
Major	File Missing in the File Server	This alarm will be reported to the system when a particular voice prompt file is not found in the IVR Prompt Server. This alarm applies only to the system with enabled IVR functionality.
Major	Switch to other RADIUS server	Appears when the current RADIUS server stops responding after three consecutive calls end in timeouts and another RADIUS server is configured, the Tenor will then switch to the next RADIUS server.
Major	Border Element connection lost.	Indicates that the Gatekeeper lost connection with the Border Element. This is cleared if the connection is re-established.
Minor	Call Event(s) Lost	A call has failed.
Minor	Missing or Incorrect Profile	The configuration profile has caused a problem.
Minor	Minor Software Error	A software error has occurred but will not affect the operation of the complete system.
Minor	Remote end did not back off in a glare situation	An incoming and outgoing call went through at the same time, and the remote end call did not back off.
Minor	Unit resource constrained	A shared resource in the unit loads the system.
Minor	Hardware component failed	A hardware component has failed. Check all components, hardware connections, etc.

Severity (appears as part of severity field)	Alarm Description (text appears in desc field)	Definition
Minor	Log RADIUS server error	Displayed when the RADIUS server fails to send required data or the data sent by the RADIUS server has improper values. Incorrect information may contain the following: RADIUS Server: Credit amount (-1) RADIUS Server: Credit minus amount RADIUS Server: Not supported currency RADIUS Server: Credit time (-1) RADIUS Server: Credit time < 6 sec RADIUS Server: Invalid error code
Informational	Gatekeeper status	Reports the status of the Gatekeeper.
Informational	Miscellaneous information	Miscellaneous information about the unit is reported. The contents of this alarm will vary.
Informational	Info Software Error	Indicates information about miscellaneous software error. This does not affect system operation.
Informational	Glare occurred	An incoming and outgoing call went through at the same time, and the remote end call did not back off, but the situation was corrected.

Display all Alarms

You are able to display both active alarms and an alarm history as follows:

1. Through CLI, access the *Monitor* prompt.

2. Type *alarm.* Both active alarms and the alarm history will be displayed. See section *How to Read Alarms* for field definitions.

Figure 5-1 Alarm sample

IP# Sequence#Type#Severity# Desc# Slot# Device# Analog Interface# Channel# Date/Time

192.166.28.230:944:ALR:3:Border Element connection lost:0:0:0:0:TUE OCT 14 14:05:27 2003

192.166.28.230:946:RPT:4:Gatekeeper status (Gatekeeper(0.0.0.0) removed):0:0:0:0:TUE OCT 14 14:05:27 2003

Display Active Alarms

You are able to display all active alarms as follows:

1. Through CLI, access the *Monitor* prompt.

2. Type *alarm a.* The active alarms will be listed. See section *How to Read Alarms* for field definitions. If you enter *alarm* without a command following it, both active alarms and the alarm history will be displayed.

Figure 5-2 Active Alarm Sample

IP# Sequence#Type#Severity# Desc# Slot# Device#Analog Interface# Channel# Date/Time

192.166.28.230:944:ALR:3:Border Element connection lost:0:0:0:0:TUE OCT 14 14:05:27 2003

Display Alarm History

1. Through CLI, access the *Monitor* prompt.

2. Type *alarm h.* An alarm history will be displayed. See section *How to Read Alarms* for field definitions. If you enter *alarm* without a command following it, both active alarms and the alarm history will be displayed.

Figure 5-3 Alarm History Sample

IP# Sequence#Type#Severity# Desc# Slot# Device# Analog Interface# Channel# Date/Time

192.166.28.230:944:ALR:3:Border Element connection lost:0:0:0:0:TUE OCT 14 14:05:27 2003

192.166.28.230:945:RPT:4:Gatekeeper status (Gatekeeper(192.168.20.175) removed):0:0:0:0:TUE OCT 14 14:05:27 2003

192.168.20.175:946:RPT:4:Gatekeeper status (Gatekeeper(0.0.0.0) removed):0:0:0:0:TUE OCT 14

Verify Unit Provisioning

An error with provisioning the *Tenor AS* may cause a number of problems. It may be a simple error, such as an incorrect IP address or telephone number. See the *Command Line Interface (CLI*) guide or the *Tenor Configuration Manager/Tenor Monitor User's Guide* for provisioning information.

Maintenance Procedures

The following are advanced maintenance procedures you can do through the *Command Line Interface (CLI)* or through the *Tenor Monitor.* See the applicable user documentation you received with the unit for detailed information.

NOTE: The instructions below are given for the *Command Line Interface (CLI)*. You can also accomplish the tasks through the *Tenor Monitor.* See the user documentation you received with the unit for detailed information.

Restore Factory Defaults

A factory default is a default setting that is configured in the factory before the unit is shipped. We do not advise that you reset the system back to factory defaults unless advised to do so or if your system becomes corrupted.

If you choose to restore the factory defaults, all current configuration will be deleted, including your password. Restoring factory defaults does not change or delete the Tenor's IP address, subnet mask or default gateway; you will still be able to communicate with the unit after a reset.

You can set all system configuration settings back to their factory defaults through the *Command Line Interface (CLI)* as follows:

1. Access the CLI through a Telnet session. See the *Command Line Interface (CLI*) guide for more information.

2. Access the *Config-VOIPNetwork-1* prompt.

3. Type *setfactory.* You will be asked if you are sure you want to set the unit back to factory defaults.

4. Type **yes** to confirm (type no to cancel the restore).

Reset System

The Reset system feature enables you to reset the system, including hardware and software. You can reset the system through the unit's Back Panel or through the *Command Line Interface (CLI).*

Back Panel. Use a blunt, thin object to press in the *Reset* button, located on Tenor's front panel. The system will reset.

Command Line Interface (CLI). You can reset the system through the *Command Line Interface (CLI)* as follows:

1. Access the CLI through a Telnet session. See the *Command Line Interface (CLI*) guide for more information.

2. Access the *Maintain-MasterChassis-1#* prompt.

3. Type *reset*. You will be asked if you are sure you want to set the unit back to factory defaults.

4. Type **yes** to confirm (type **no** to cancel the restore).

Change Password

For security purposes, you may want to change the password. You can change the password through the *Command Line Interface (CLI)* as follows:

1. Access the CLI through a Telnet session. See the *Command Line Interface (CLI*) guide for more information.

2. Access the *Maintain#* module.

3. Type *password*. A prompt will ask you for the old password.

4. Type the old password and press **Enter.** A prompt will ask you for the new password. Type the new password and press **Enter.** A confirmation will ask you to confirm the new password.

5. Re-type the new password and press **Enter**.

A message will tell you the password was changed successfully.

Change Unit Date and Time

You can change the unit's date and time via *Command Line Interface (CLI)* as follows:

1. Access the CLI through a Telnet session. See the *Command Line Interface (CLI*) guide for more information.

2. Access the *Config#* module.

3. Type *date* followed by *mm/dd/yy/hh:mm:ss* and press **Enter**.

4. For example, type **config# date 06/14/02/22:14:00.** This command will set the current time to June 14, 2002 at 10:14 p.m.

If you need Additional Help

If you suspect the problem to be on the network end, contact your Central Office to verify proper operation.

After completing all troubleshooting/maintenance procedures and reviewing the Common Symptoms/Problems section, you can contact the Customer Service Department at the following:

Quintum Technologies, Inc.
71 James Way
Eatontown, NJ 07724
For domestic calls: (877) 435-7553
For international calls: (732) 460-9399
email: service@quintum.com

Chapter 6: Advanced Topic: SNMP/IVR

This chapter explains advanced topics for using the unit, including SNMP and IVR.

- *SNMP Description*
- *Install SNMP*
- *Working with SNMP*
- *IVR*
- *Typical IVR Network Connection*
- *Configure IVR Voice Prompts*
- *IVR Call Flow Specifications*

SNMP

Simple Network Management Protocol (SNMP) is the standard protocol used to exchange network management information. It is managed by Hewlett Packard®'s HP Openview™ Network Node Manager. Specifically, this chapter tells you how HP® OpenView auto-discovers a *Tenor AS* unit, as well as generate SNMP traps for existing alarm messages.

SNMP management requires two primary elements: a network manager and an SNMP agent. A network manager is the software running on a workstation through which the network administrator monitors and controls the different hardware and software systems that comprise a network. The agent is a piece of software running on network equipment that implements the SNMP protocol. SNMP defines exactly how a network manager communicates with an SNMP agent. For example, SNMP defines the format of request that a network manager sends to an agent and the format of replies the agent returns.

Through an SNMP management system, the network manager can communicate and manage several different network devices at the same time. The network manager polls the SNMP agents (such as routers, hubs or network servers) for certain requested information. The agent will then gather information about the machine it is running on and carries requests from the network manager to read and change the information.

How does *Tenor AS* utilize SNMP?

The *Tenor AS* unit supports the SNMP protocol: specifically, Hewlett Packard®'s HP Openview™ software as the SNMP network manager for the *Tenor AS* unit. Once you set up HP Openview to view and "auto-discover" *Tenor AS* as a network device using SNMP, HP Openview will be able to issue commands, get responses, and perform certain functions. For example, you can configure the SNMP agent in the *Tenor AS* to generate and send traps for existing *Tenor AS* alarms to HP Openview.

A *Tenor AS* unit can report alarms for up to three network managers.

Installation Requirements

You will need to install and run HP Openview NMS 6.0 in order to recognize the *Tenor AS* as an SNMP agent. Below are basic hardware and software requirements you will need to install HP Openview. See your HP Openview documentation for detailed information and installation instructions.

Minimum hardware requirements

- Intel Pentium 120 Mhz processor-based computer
- CD Drive
- 96 MB RAM
- 250 MB free disk space on one drive
- 80 MB free page filing space

Software requirements

- Microsoft Windows NT operating system (version 4.0), with TCP/IP version 4.0
- Microsoft SNMP Agent (SNMP Services)
- Microsoft Peer Web Services or Internet Information Server (IIS) version 4.0
- Netscape Navigator web browser (version 4.06 or higher) or Microsoft Internet Explorer web browser (version 4.0 or higher) with Java/JavaScript options enabled.

Install SNMP

There are two steps you need to accomplish before HP Openview can interact with a *Tenor AS* unit as an SNMP agent.

- Download and install HP Openview configuration files specific for *Tenor AS*
- Configure the IP address of the network manager in the *Tenor AS*

Download and install SNMP-Related Files

For HP Openview (network manager) to view and manage the *Tenor AS* as an SNMP agent, you must first download and install files from the *www.quintum.com* web site as follows:

1. Start up HP Openview. (For specific information, see the user documentation you received with the software.) There are two ways to download the applicable files: from the *www.quintum.com* web site or from the CD delivered with the system. To download from the web, go to step 2. Otherwise, for CD installation, go to step 5.

2. From the web site, access a web browser (i.e., Microsoft Internet Explorer, Netscape) and go to the web site *www.quintum.com* and access **Customer Service/ Training**. If you are not registered, a registration form will be displayed. Once registered, you will be able to download the appropriate file. See Step 3.

3. Download the *SNMP-HPOV-Plugin.zip* file to your PC.

4. Unzip the *SNMP-HPOV-Plugin.zip* file to the HP Openview root directory (this is the directory in which HP Openview was installed).

   The following files will be listed:

installquintum.bat	quintum.reg	quintum.exe
quintum.conf	quintum.fields	quintum_type
connector_qdvoip	connector_qavoip	uninstallquintum.bat
quintum_sym	Tenor_chassis.ico	Tenor_A.ico
Tenor_D.ico		

5. From the Quintum CD ROM, click on **TenorTools**. Download the **SNMP Agent Software** to the HP Openview root directory (this is the directory in which HP Openview was installed).

6. From the HP Openview root directory run *installquintum.bat.* Installation will begin. The following section lists where the installation files are copied and which files were modified. You can use this information to make changes manually to files, if desired.

NOTE: If you have already run the *installquintum.bat*, you must first run the *installlquintum.bat clean* command prior to reinstalling. This command will restore any changes made to the quintum files back to the default parameters (see below for which files are modified upon installation).

- These lines are added to the file *HPOVRoot\conf\oid_to_sym*:

```
1.3.6.1.4.1.6618.1.1.1:Connector:QDVOIP       # QUINTUM Digital Tenor
1.3.6.1.4.1.6618.1.1.2:Connector:QAVOIP       # QUINTUM Analog Tenor
1.3.6.1.4.1.6618.1.1.10:Connection:QCVOIP     # QUINTUM Chassis Tenor
```

- These lines are added to the file *HPOVRoot\conf\oid_to_type*:

```
1.3.6.1.4.1.6618.1.1.1:Quintum:Epilogue:H   # Quintum-Digital Tenor
1.3.6.1.4.1.6618.1.1.2:Quintum:Epilogue:H   # Quintum-Analog Tenor
1.3.6.1.4.1.6618.1.1.10:Connection:Epilogue:H #QUINTUM Chassis Tenor
```

- These lines are added to the file *HPOVRoot\conf\C\trapd.conf* using the "xnmevents -load" HPOV command:

```
OID_ALIAS Quintum .1.3.6.1.4.1.6618
#
#
#
EVENT Critical_Alarm .1.3.6.1.4.1.6618.2.0.1 "Status Events" Critical
FORMAT Received trap:generic #$G specific #$S. $#args:$*
SDESC
T1IU_in_loopback;_network_request
EDESC
#
#
#
EVENT Major_Alarm .1.3.6.1.4.1.6618.2.0.2 "Status Events" Major
FORMAT Received trap:generic #$G specific #$S. $#args:$*
SDESC
Corruption_of_configuration_info.
EDESC
#
#
#
EVENT Minor_Alarm .1.3.6.1.4.1.6618.2.0.3 "Status Events" Minor
FORMAT Received trap:generic #$G specific #$S. $#args:$*
SDESC
RIU_not_installe
EDESC
```

```
#
#
#
EVENT Report_Status_Alarm .1.3.6.1.4.1.6618.2.0.4 "Status Events" Normal
FORMAT Received trap:generic #$G specific #$S. $#args:$*
SDESC
Uswcomment
EDESC
#
#
#
EVENT Uswcomment .1.3.6.1.4.1.6618.2.0.0 "Status Events" Normal
FORMAT Received trap:generic #$G specific #$S. $#args:$*
SDESC
Uswcomment
EDESC
```

- The *quintum.exe* file is added to the HPOVRoot\bin directory.

- These files are added to the *HPOVRoot\bitmaps\C* directory:

  Tenor_chassis.ico
  Tenor_A.ico
  Tenor_D.ico

- The *quintum.fields* file is added to the HPOVRoot\fields\C directory, and is installed and verified using the "OVW -fields" command.

- The *quintum.reg* file is added to the HPOVRoot\registration\C directory.

- These files are added to the D:\hpovnnm\symbols\C\Connector directory:

  CONNECTOR_QDVOIP
  CONNECTOR_QAVOIP
  CONNECTOR_QCVOIP

7. HP Openview will then identify and “auto-discover” the Tenor unit as an SNMP agent. An icon will appear on the screen to represent each *Tenor AS* unit. See below.

Figure 6-1 Tenor SNMP Agent Icon

Configure Network Manager IP address

Through *Tenor AS*’s *Command Line interface (CLI)*, you can configure the IP address for the network management site where HP Openview is running. Once the IP address is configured, the *Tenor AS* will be able to process and generate traps for existing alarms, which will enable HP Openview to monitor the *Tenor AS* unit for alarm states.

To set the IP address, use the following CLI command:

con-MasterChassis> snmptrapip1 <ip>. This command configures the IP address of the network management site in the *Tenor AS* unit from which the SNMP traps will be generated. Valid entry: The index number of 1-3 (the index number is used to designate the index # for the network manager to be used) followed by the IP address for the network management site. You can assign a *Tenor AS* unit up to three network managers; each manager will have a different index number.

Assign an IP address as follows:

NOTE: The following instructions assume your PC is running Windows 95 or later.

1. Click on *Start> Run.* The *Run* window will appear.

2. Type *telnet* in the Open box and click on **Ok**. The Telnet GUI will launch.

3. Click *Connect> Remote System.*

4. In the *Host Name* window, enter the IP address of the unit from which you would like to view alarms.

5. In the port window, choose telnet (port 23 is the default port for a telnet session).

6. From the *TermType* drop down list, select *vt100*.

7. Click **Connect.** A telnet session will be displayed. (To exit a telnet session at any time, type *exit* at the prompt.)

8. Enter the same password you set via CLI.

9. At the c*onfig - MasterChassis prompt*, type *set snmptrapip1 <ip>*. For example, type *set snmptrapip1 208.226.140.12.* This will assign index of 1 to indicate the first network manager being assigned to that *Tenor AS* unit and assign the IP address 208.226.140.12 as the network manager. To assign the second snmptrap ip, type *set snmptrapip2 <ip>* and the third type set *snmptrapip3 <ip>*.

10. Repeat step 9 for each network manager (up to 3).

11. Type **submit**. The new IP address(es) will be submitted to the applicable *Tenor AS* units.

Working with SNMP

View traps

You can view the traps HP Openview received from the *Tenor AS* unit as follows:

NOTE: Ensure HP Openview is running.

1. From the HP Openview map where the *Tenor AS* units appear, right-click on the *Tenor AS* icon for which you want to view traps and select *Alarms*.

2. From the *Alarm Categories* window, click on *All Alarms*. All traps (alarm messages) will be listed in the order in which they occurred.

View Alarm Status via *Tenor AS* icon

Through the color of a *Tenor AS* icon on the HP Openview desktop, you can determine the alarm state of the unit as well as view the corresponding alarms. Valid icon colors are listed below.

- Green Icon. No Alarm indicated.
- Red Icon. Critical Alarm (used for when HP Openview cannot communicate with the *Tenor AS* or there is a severity level 1 alarm).
- Orange Icon. Major Alarm (used for severity level 2 alarms).
- Yellow Icon. Minor Alarm (used for severity level 3 alarms).
- White Icon. Info Alarm (used for severity level 4 alarms).

View the alarms associated with a *Tenor AS* unit as follows:

NOTE: Ensure HP Openview is running.

1. From the HP Openview desktop, identify the icon for the *Tenor AS* unit from which you would like to view alarms.

2. According to the color of that icon, determine the alarm state from the list above.

3. To view the alarms associated with that icon, right-click from anywhere on the HP Openview desktop and select *Alarms.* The *All Alarms Browser* window will appear, which contains the list of alarms.

Launching Command Line Interface (CLI) from HP Openview

Launch Command Line Interface with pop-up menu as follows:

1. From the HP Openview desktop, right-click on the desired *Tenor AS* unit icon and select *Quintum Analog.*

2. The CLI will launch with the IP address of the unit associated with that icon (the correct IP address will appear in the URL).

3. Configure the unit.

OR

Launch CLI by double-clicking the icon as follows:

1. From the HP Openview desktop, double-click on the desired *Tenor AS* unit icon and select *Quintum Analog.*

2. The CLI will launch with the IP address of the unit associated with that icon (the correct IP address will appear in the URL).

3. Configure the unit. See the *Tenor Configuration Manager/Tenor Monitor* User's Guide for more information.

Set up *Tenor AS* status polling

The following explains how to configure HPOV to determine and display unit status if the *Tenor AS* is not configured to send traps. Polling is only intended for *Tenor AS* units which are not accessible to configure the network manager IP address using the *Tenor AS*'s *Command Line Interface (CLI).* We do not recommend polling more than 10 *Tenor AS* units.

1. From the HP Openview desktop, right-click on the desired *Tenor AS* unit icon and select *Object Properties.*

2. *From the Attributes* window, select *Quintum Tenor AS* attributes.

3. Click on *Edit Attributes.*

4. Select *Turn on Tenor Status Polling* (the toggle allows you to set to True or False). Select True.

5. Click on **Verify**. After the verification is complete, click **Ok.**

Polling occurs in 30 second intervals, or whenever a *Tenor AS* trap is received.

Set up Debug Message Display window

The following explains how to display debug messages that are generated by the *quintum.exe*. This window is used to troubleshoot HPOV processing events for the *Tenor AS* icons.

1. From the HP Openview desktop, right-click on the desired *Tenor AS* unit icon and select *Object Properties.*

2. From the *Attributes* window, select *Quintum Tenor AS.*

3. Click on *Edit Attributes.*

4. Select *Turn on Debug Window* (the toggle allows you to set to True or False). Select True.

5. Click on **Verify**. After the verification is complete, click **Ok.**

The Debug Message Display window displays debug messages being processed that are associated with that icon. Only one window can be opened at a time.

IVR

Interactive Voice Response (IVR) is a feature of the *Tenor AS* that enables you to offer services, such as Pre-paid calling cards and Post-paid accounts to your customers. The *Tenor AS* uses the RADIUS (Remote Authentication Dial-In User Service), for authenticating and authorizing user access to the VoIP network. The RADIUS is a standard protocol which provides a series of standardized messages formats for transmitting and receiving dialed information, account data and authorization codes between the network access gateway and the billing server. As a result, the RADIUS enables the *Tenor AS* to interoperate directly with billing server application software from a wide range of vendors. To provide redundancy, the *Tenor AS* supports two RADIUS servers: Primary and Secondary.

The IVR interface enables the *Tenor AS* to play back interactive pre-recorded voice messages to a customer calling in from the Public Switched Telephone Network (PSTN), requesting information such as account number, PIN number, and calling number. The caller is prompted for each piece of information and the digits are captured by the *Tenor AS* and converted into RADIUS format. The RADIUS will use the input data to identify the customer, verify the identity using the PIN code, check the account status, and then send back messages in RADIUS format to authorize the *Tenor AS* to proceed with the call. The call will then be routed over the VoIP network to the appropriate remote *Tenor AS*.

You are able to pre-record and customize voice prompt files which lead the customer through the calling card procedure. You can pre-record messages to meet your network and customer needs. Seven languages are supported for voice prompts: English, French, Mandarin, Spanish, German, Arabic, or Persian. The multiple language feature enables you to select multiple languages in which to play voice recordings. For example, the welcome message will be played in Persian, and the following message will ask the user if they would like to hear the rest of the messages in English, French, Mandarin, Spanish, German, Arabic, or Persian.

Through the *Command Line Interface (CLI)*, you can configure all functions within the *Tenor AS* that will be used for pre-paid/post-paid call services, including IVR and RADIUS data.

IVR Call Types

There are two call services *Tenor AS* supports through IVR: Pre-paid Calling Card and Post-paid Account.

- **Pre-paid Calling Card**. A card with a set dollar amount to be used for placing voice calls. The normal call flow for a pre-paid service starts with the caller dialing the pre-paid access number from a regular phone. The *Tenor AS* prompts the caller for a card number and destination number which are sent to the RADIUS server (third party software) for verification.
- **Post-paid Account.** An account used for placing voice calls where the account number and PIN (personal identification number) are used for security. The *Tenor AS* prompts the caller to enter an account number and a PIN (personal identification number), which are sent to the RADIUS server for verification. Normally, there is no limit on the amount or duration the caller is authorized to talk.

ANI Authentication

There are two types of ANI Authentication supported in the *Tenor AS*: Type 1 and Type 2.

ANI Authentication Type 1 enables calling subscribers to receive authentication based on the calling number. If an incoming call has no associated ANI information, the call will not be answered, but simply disconnected. If ANI is present, it will be authenticated with the Radius server (before answering the call). If authentication fails, the call will not be answered but simply disconnected. If authentication succeeds, the call will be answered and second dial tone will be provided.

ANI Authentication Type 2 enables calling subscribers to receive three authentication types based on the calling number: (1) If there is an ANI in the setup message, authentication with the ANI will be done (2) If there is no ANI information in the setup message, the user will be prompted for a PIN number and (3) If the incoming packet has the ANI, but authentication with the ANI fails, the user will be prompted for a PIN number. If authentication succeeds, a 2nd dial tone is sent to receive a DNIS. If there is ANI information in incoming packet but the authentication with the ANI fails, the caller is prompted for a PIN number. If the authentication succeeds, a 2nd dial tone is sent to receive a DNIS.

Multi-session

Multi-session enables the user to make another call at the end of a call, without having to hang up and call again. At the end of the first call, a voice prompt will ask if the user wants another call. In addition, the user will be able to terminate an ongoing call and make another call, by pressing ** or ##.

Typical IVR Network Connection/Process

See Figure 6-2 for a typical network configuration for IP based pre-paid calling card/post-paid account services. This figure illustrates the originating *Tenor AS* is configured to provide pre-paid/ post-paid services. All the subscribers will have to dial into this *Tenor AS* via PSTN to get access to pre-paid/post-paid services.

Each process is denoted by a number (in bold); each number and the corresponding process is explained in Table 6-1.

Figure 6-2 Typical Network Connection/Call Flow Process

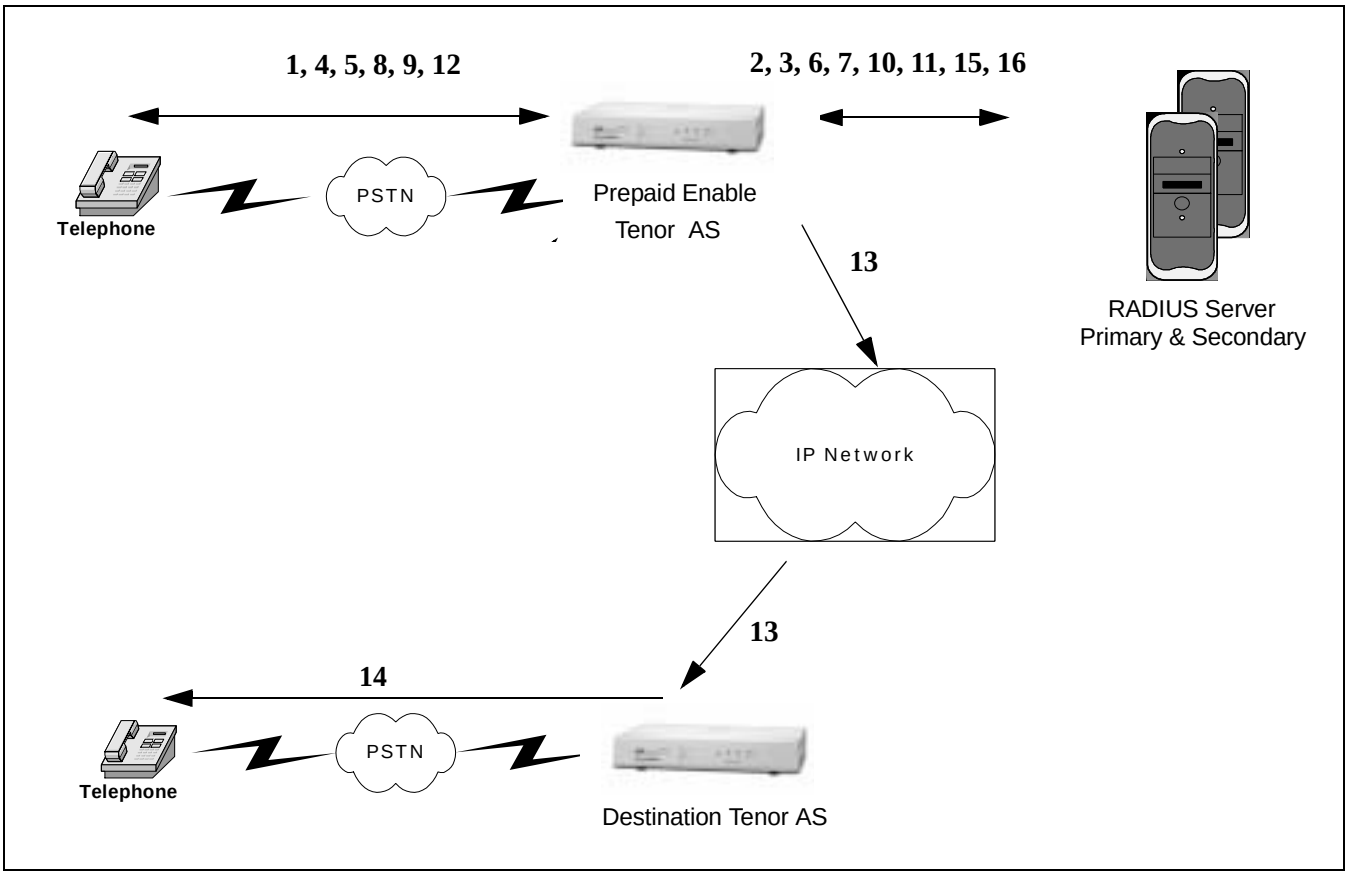


Table 6-1 Call Flow Process for Figure 9-1

Number	Process
1	User dials access number to initiate call (e.g., 1-800-xxx-xxxxx).
2	*Tenor AS* answers call and determines user is calling a pre-paid service based on the number dialed or the trunk group to which the call is transferred. *Tenor AS* sends Start Accounting to RADIUS server. Start Accounting is optional.

3	RADIUS server acknowledges start accounting request. Start Accounting is optional.
4	*Tenor AS* prompts the subscriber for the card number.
5	Caller enters the card number.
6	*Tenor AS* sends the card number to the RADIUS server for verification.
7	RADIUS server returns the credit balance for the account.
8	*Tenor AS* plays the credit amount and prompts the subscriber for the destination number.
9	Caller enters the destination number.
10	*Tenor AS* sends the destination number to the Radius server for authorization.
11	RADIUS server responds with the duration for which the caller is authorized to speak to the destination number.
12	*Tenor AS* plays the authorized duration.
13	The call is sent via IP network to the destination *Tenor AS*.
14	The destination *Tenor AS* sends the call via PSTN to the destination phone and the call is established.
15	If either side disconnects, *Tenor AS* sends stop accounting request to the RADIUS server.
16	RADIUS server acknowledges the stop accounting request.

Configure IVR - Quick Start

For a *Tenor AS* to be used as part of the complete IVR system, and communicate effectively with the RADIUS, you must configure basic options in the *Tenor AS* via *Command Line Interface (CLI)* for the following:

- Basic IVR Data
- RADIUS

Instructions for configuring basic IVR information via *Command Line Interface (CLI)* are described below. For detailed information about all CLI commands available for IVR and detailed information about the command described below, see the *Command Line Interface* guide you received with the unit.

Basic IVR Data (via Trunk Group)

Configure basic IVR information as follows:

1. Access CLI.
2. Access the *config-trunkcircuitroutinggroup-1#* prompt.
3. Type *set ivrtype 2* (sets the IVR type to pre-paid calling services).
4. Type *set ivraccessnumber,* followed by the telephone number used to access the IVR system (up to 14 digits).
5. Type *ivrcardlength 10* (10 is an example, the length of the calling card is numeric up to 20 digits).
6. Access the applicable *config- trunkcircuitroutingroup - 1#* prompt.
7. Set the *ivrlanguage* field to the applicable language: 1 (English), 2 (Persian), 3 (Mandarin), or 4 (French), 5 (Spanish), 6 (German), or 7 (Arabic).

RADIUS Server

Configure RADIUS data via *config radius#* prompt as follows:

1. Access CLI.
2. Access the *config-radius-1#* prompt.
3. Type *set psipa,* followed by the IP address for the primary RADIUS server (i.e., *set psipa 208.22.234.34*). The “p” indicates the primary RADIUS server.
4. Type *set ssipa* followed by the IP address for the secondary RADIUS server (i.e., *set ssipa 208.22.234.32*). The “s” indicates the secondary RADIUS server.

5. Type *sharedsecret* (sharedsecret is similar to password), followed by the RADIUS key (up to 63 characters) (i.e., *sharedsecret 454AJU)*.

6. Ensure *accountingtype* is configured. If this field is set to 0, no “stop accounting” messages will be sent.

Configure IVR Voice Prompts

A voice prompt is a pre-recorded message played for a caller at specific times during a call; the actual voice messages used in the prompting process is stored in the system controller card in the form of audio files that can be recorded by a recording studio. The audio files can then be played back over the PSTN connection to the caller in messages such as " Enter Account Number", "Enter PIN number", "Enter Card Number", etc.

When a *Tenor AS* system is re-started, it has all voice prompts stored in the *Tenor AS*'s cache memory for replay.

Voice Prompt Requirements (English Requirements)

NOTE: To ensure good speech quality, the voice prompts should be professionally recorded in a studio.

Table 6-2 specifies requirements for each voice file recorded. Definitions for each column entry are listed below.

- **Filename.** Actual filename under which the appropriate pre-recorded speech is to be stored on the system controller card. The voice files should be in ***CCITT G711 μ-Law*** format and stored with ***.wav*** extension (no application-specific headers, like in the files generated by the Microsoft Recorder, are allowed).
- **Suggested Content.** This field contains a particular piece of speech which represents the voice file, played back along with other file(s) during the actual IVR call. For these particular cases, the voice files should be carefully recorded, using appropriate intonation patterns.
- **Example with other Voice Files.** This field uses the specified voice file with other possible voice files. For example, the file [enter] + [account_number] has the following contents: Please enter your account number. The "+" indicates "in addition to" the filename.
- **Usage Example.** This field represents the actual pre-recorded speech being used in a specific speech pattern.

Table 6-2 File Conventions for Recorded Files

Filename	Suggested Content	Example with other Voice Files	Usage Example
account_expired	**"Your card has expired."**	[account_expired] + + [good_bye]	Your card has expired. Goodbye.
account_number	**"account number"**	[enter] + [account_number] + [pound_key]	Enter account number followed by the pound key.
and	**"and"**	-	-

blocked	**"The number you have dialed is blocked."**	[blocked] + [hang_up]	The number you have dialed is blocked. Please hang up and call again later.
card_number	**"card number"**	[enter] + [card_number] + [pound_key]	Please enter your card number followed by the pound key.
cent	**"cent"**	-	-
cents	**"cents"**	-	-
credit_limit	**"Your credit limit is exceeded."**	[credit_limit] + [good_bye]	Your credit limit is exceeded. Good bye.
destination	**"destination number"**	[enter] + [destination] + [pound_key]	Please enter your destination number followed by the pound key.
dollar	**"dollar"**	-	-
dollars	**"dollars"**	-	-
eight	**"eight"**	-	-
eighteen	**"eighteen"**	-	-
eighty	**"eighty"**	-	-
eleven	**"eleven"**	-	-
enter	**"Please enter your"**	[enter] + [card_number OR account_number OR pin OR destination]	Please enter your card number. Please enter account number. Please enter pin. Please enter destination number.
fifteen	**"fifteen"**	-	-
fifty	**"fifty"**	-	-
five	**"five"**	-	-
forty	**"forty"**	-	-
four	**"four"**	-	-
fourteen	**"fourteen"**	-	-
good_bye	**"Good bye."**	-	-

hang_up	**“Please hang up and call again later.”**	-	-
hundred	**“hundred”**	-	-
in_use	**“This account is currently in use.”**	[in_use] + [hang_up]	This account is currently in use. Please hangup and call again later.
invalid	**“You have entered an invalid number.”**	-	-
language	**“for English press”**	[lang] + [one]	For English press one.
minute	**“minute”**	-	-
minutes	**“minutes”**	-	-
next call_ pound	**“To make another call press the star key twice”**	-	-
next call_star	**“To make another call press the pound key twice”**	-	-
nine	**“nine”**	-	-
nineteen	**“nineteen”**	-	-
ninety	**“ninety”**	-	-
no_funds	**“You have insufficient funds in your account.”**	[no_funds] + [good_bye]	You have insufficient funds in your account. Good bye.
one	**“one”**	-	-
pin	**“pin”**	[enter] + [pin] + [pound_key]	Please enter your pin followed by the pound key.

pound_key	**"followed by the pound key."**	[enter] + [account_number OR card_number OR pin OR destination] + [pound_key].	Please enter your account number followed by the pound key. Please enter your card number followed by the pound key. Please enter your pin followed by the pound key. Please enter your destination number followed by the pound key.
problems	**"We are currently experiencing technical difficulties."**	[problems] + [hang_up]	We are currently experiencing technical difficulties. Please hang up and call again later.
remaining	**"remaining."**	[<money> OR <time>] + [remaining]	Eighteen minutes remaining. Twenty cents remaining.
rial	**"rial"**	-	-
second	**"second"**	-	-
seconds	**"seconds"**	-	-
seven	**"seven"**	-	-
seventeen	**"seventeen"**	-	-
seventy	**"seventy"**	-	-
six	**"six"**	-	-
sixteen	**"sixteen"**	-	-
sixty	**"sixty"**	-	-
ten	**"ten"**	-	-
thirteen	**"thirteen"**	-	-
thirty	**"thirty"**	-	-
thousand	**"thousand"**	-	-
three	**"three"**	-	-
twelve	**"twelve"**	-	-
twenty	**"twenty"**	-	-

two	**“two”**	-	-
wait	**“Thank you. Please hold.”**	-	-
welcome	**“Welcome!”**	-	-
yen	**“yen”**	-	-
you_have	**“You have”**	[you_have] + [<any number>] + [remaining]	You have six cents remaining.
yuan	**“yuan”**	-	-
zero	**“zero”**	-	-
zero_balance	**“You have zero balance in your account.”**	[zero_balance] + [good_bye]	You have zero balance in your account. Goodbye.

Create Voice Prompt Files

Configure the directory which houses the voice prompts as follows:

1. Access the root directory (i.e., hd/ivr).
2. Create a subdirectory for each supported language and use the following specific naming conventions:
   - For English, use the hd/ivr/English
   - For Persian, use the filename hd/ivr/Persian
   - For Mandarin, use the filename hd/ivr/Mandarin
   - For French, use the filename hd/vr/French
   - For Spanish, use the filename hd/ivr/Spanish
   - For German, use the filename hd/ivr/German
   - For Arabic, use the filename hd/ivr/Arabic
3. Record all voice prompts and store under the appropriate directory (i.e., hd/ivr/English subdirectory).

IVR Call Flow - Specifications

Pre-paid Calling Card - Call Flow (with default language)

Figure 6-3 is a diagram of the call flow for pre-paid calling card service, which details the messages transmitted between the following components:

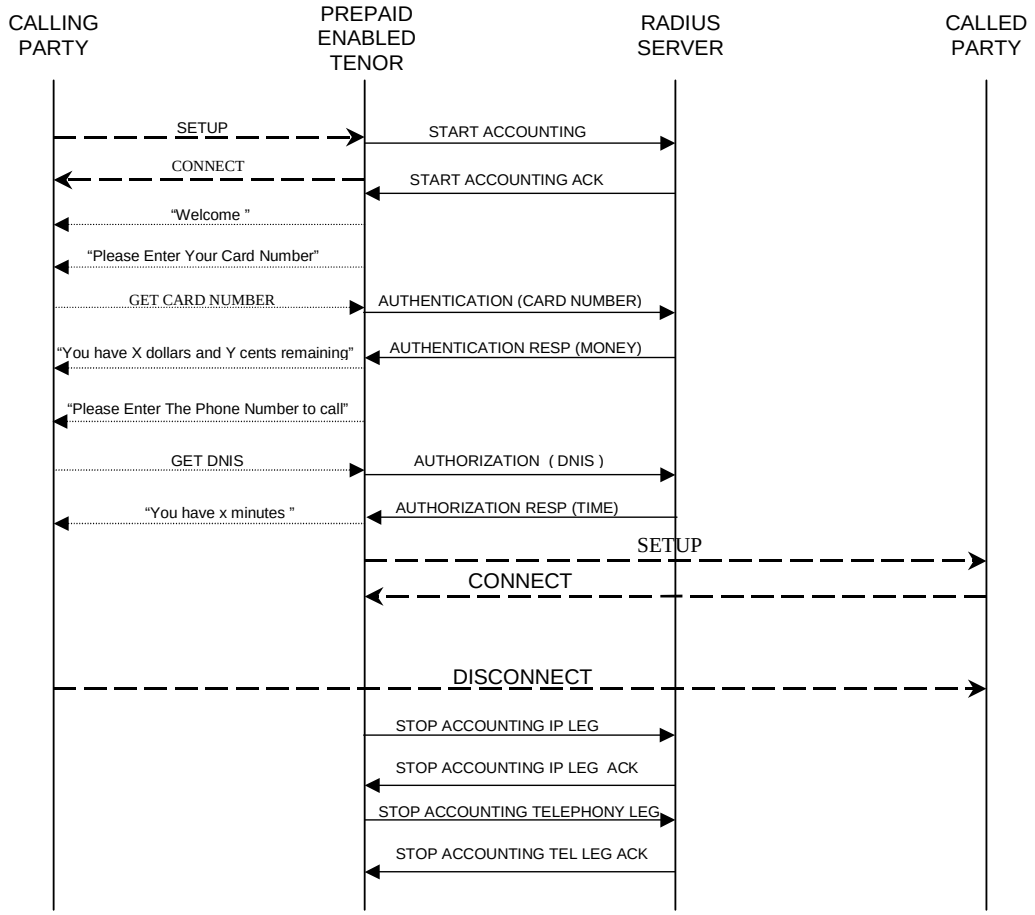
Calling Party. The originating caller using a pre-paid calling card.

Prepaid Enabled Tenor. The *Tenor AS* performing the IVR functions.

RADIUS Server. Remote Authentication Dial-In User Service for authenticating and authorizing user access to the VoIP network. The RADIUS provides a series of standardized messages formats for transmitting and receiving dialed information, account data and authorization codes between the network access gateway and the billing server.

Called Party. The destination called party.

Figure 6-3 Pre-paid Calling Card - Call Flow (default language)



Post-paid Calling Card - Call Flow (with default language)

Figure 6-4 is a diagram of the call flow for post-paid service, which details the messages transmitted between the following components:

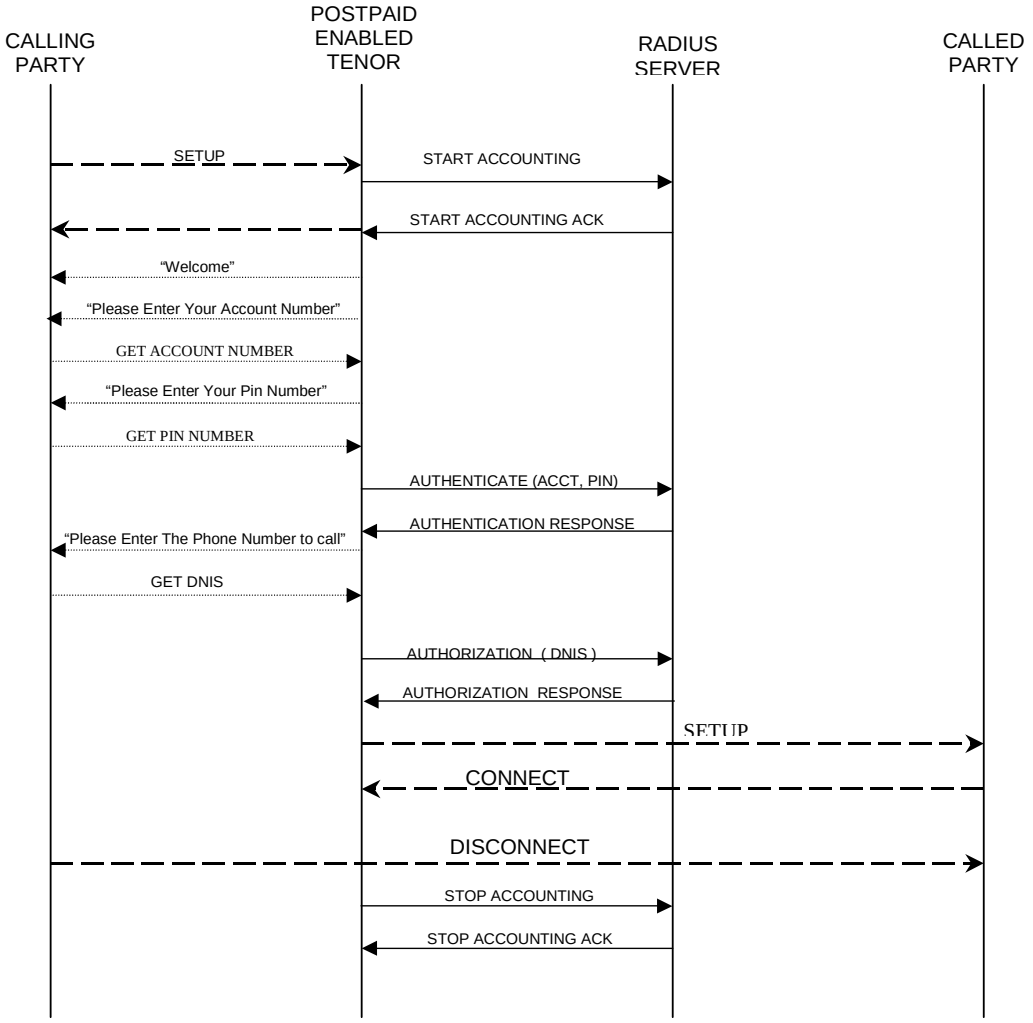
Calling Party. The originating caller using a post-paid calling party.

Prepaid Enabled *Tenor AS*. The *Tenor AS* performing the IVR functions.

RADIUS Server. Remote Authentication Dial-In User Service for authenticating and authorizing user access to the VoIP network. The RADIUS provides a series of standardized message formats for transmitting and receiving dialed information, account data and authorization codes between the network access gateway and the billing server.

Called Party. The destination called party.

Figure 6-4 Post-paid Account - Call Flow (default language)



Pre-paid and Post-paid Calling Card - Call Flow (with multiple language support)

Figure 6-5 is a diagram of the call flow for pre-paid calling card service, which details the messages transmitted between the following components:

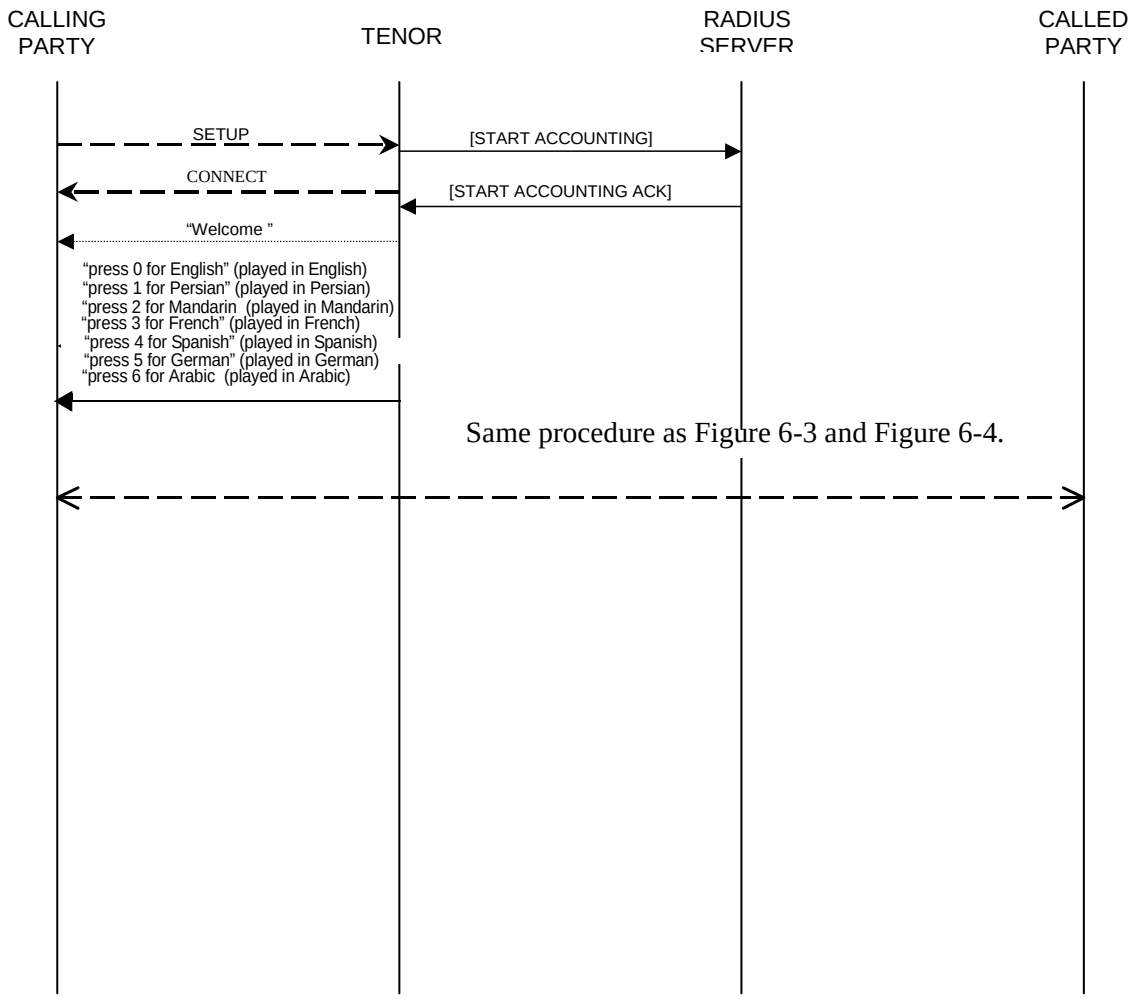
Calling Party. The originating caller using a pre-paid calling card.

Tenor. The *Tenor AS* performing the IVR functions.

RADIUS Server. Remote Authentication Dial-In User Service for authenticating and authorizing user access to the VoIP network. The RADIUS provides a series of standardized messages formats for transmitting and receiving dialed information, account data and authorization codes between the network access gateway and the billing server.

Called Party. The destination called party.

Figure 6-5 Pre-paid and Post-paid Calling Card - Call Flow (multiple language support)


CALLING PARTY
TENOR
RADIUS SERVER
CALLED PARTY
SETUP
[START ACCOUNTING]
CONNECT
[START ACCOUNTING ACK]
"Welcome "
"press 0 for English" (played in English)
"press 1 for Persian" (played in Persian)
"press 2 for Mandarin (played in Mandarin)
"press 3 for French" (played in French)
"press 4 for Spanish" (played in Spanish)
"press 5 for German" (played in German)
"press 6 for Arabic (played in Arabic)
Same procedure as Figure 6-3 and Figure 6-4.

Pre-paid and Post-paid Calling Card - Call Flow (with Multi-Session Call support)

For a multi-session call, the calling party can interrupt the call by pressing a multi-session key at anytime and making a new call. When the called party disconnects the call first, the *Tenor AS* asks if the caller wants another call; the user can then press the designated key. Figure 6-6 is a diagram of the call flow for pre-paid and post-paid call card service (with multi-session support), which details the messages transmitted between the following components:

Calling Party. The originating caller using a pre-paid calling card.

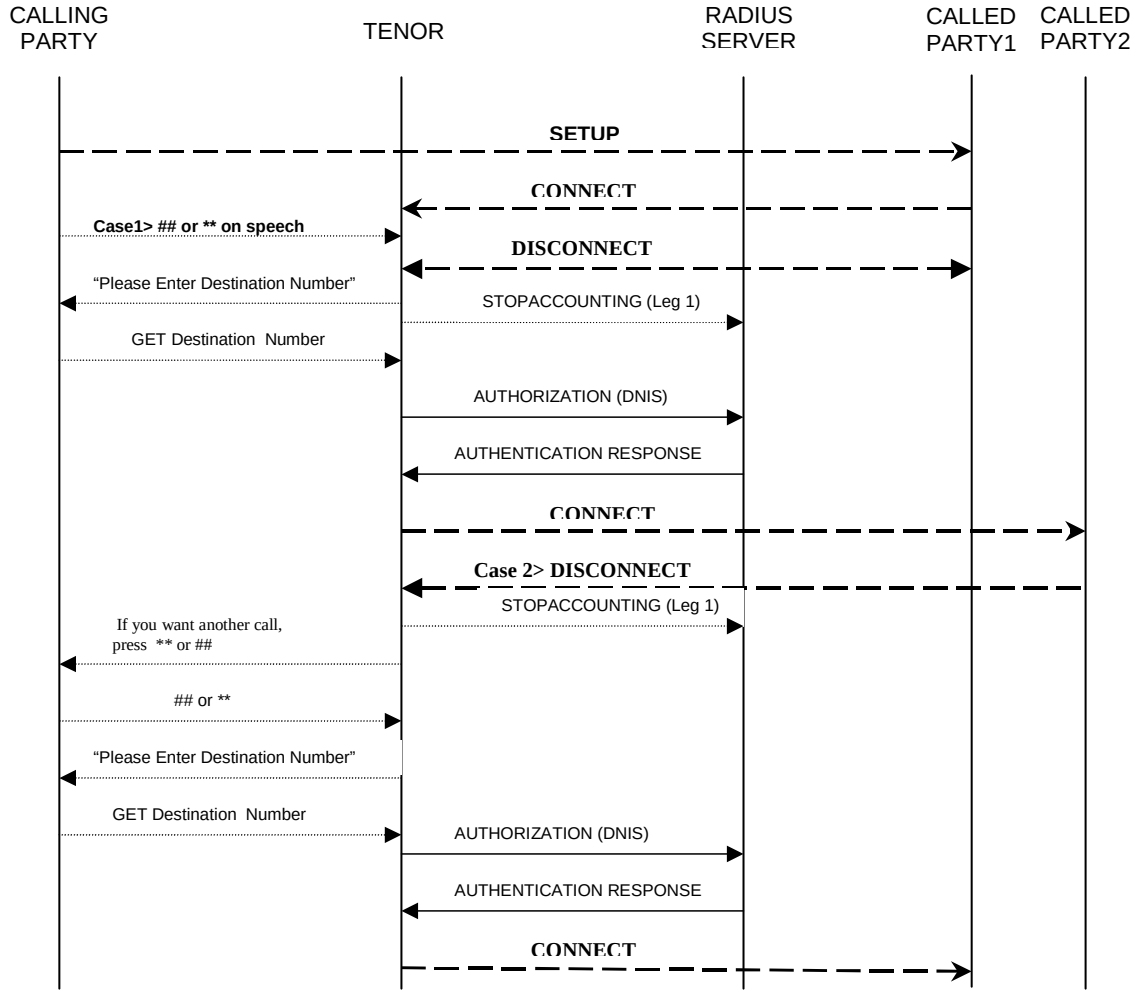
Tenor. The *Tenor AS* performing the IVR functions.

RADIUS Server. Remote Authentication Dial-In User Service for authenticating and authorizing user access to the VoIP network. The RADIUS provides a series of standardized messages formats for transmitting and receiving dialed information, account data and authorization codes between the network access gateway and the billing server.

Called Party 1. The first destination called party.

Called Party 2. The second destination called party.

Figure 6-6 Pre-paid and Post-paid Calling Card - Call Flow (multi-session support)



ANI Authentication Application Type 1 - Call Flow

ANI Authentication Application Type 1 enables calling subscribers to receive authentication based on the calling number. If you configure the *ivrtype* to 4 (ANI Type 1), when an incoming call comes in, the call will be authenticated with ANI by a RADIUS server.

Figure 6-7 is a diagram of the call flow for ANI Authentication Application Type 1, which details the messages transmitted between the following components:

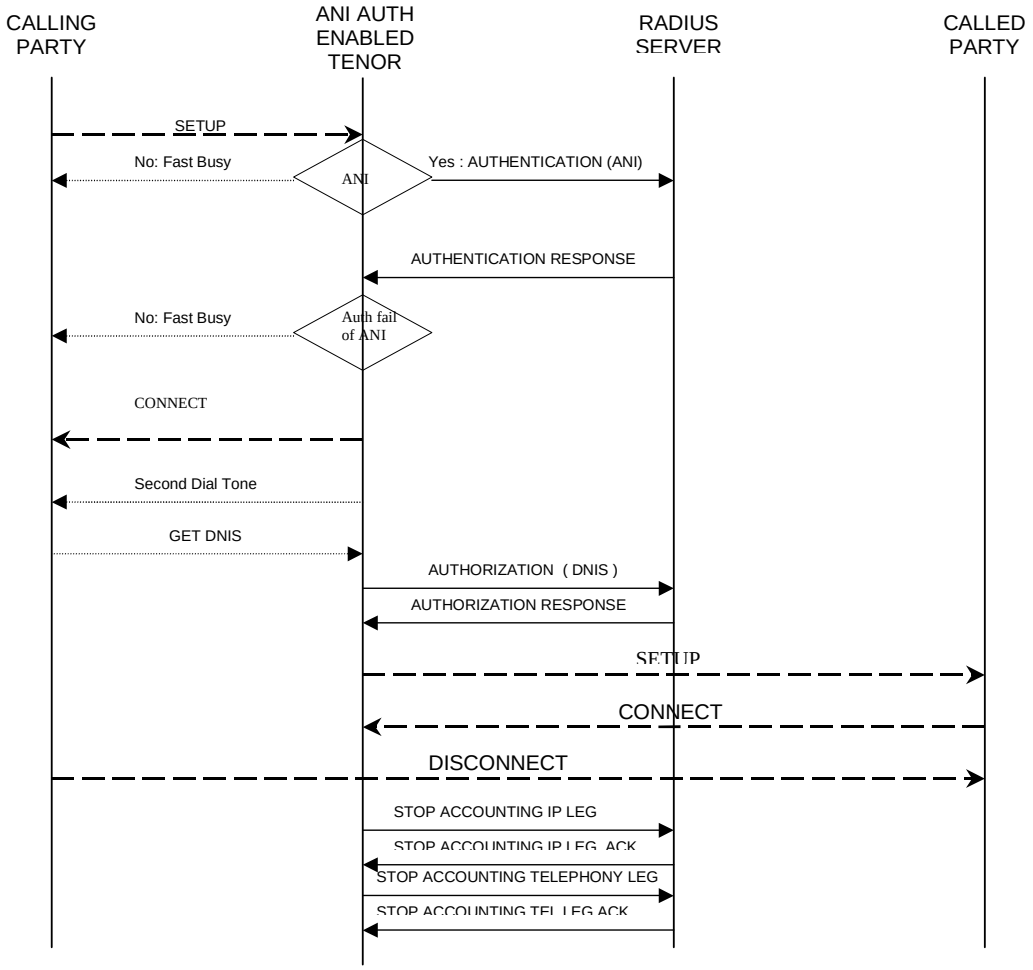
Calling Party. The originating caller using a pre-paid calling card.

ANI AUTH Enabled *Tenor AS*. The *Tenor AS* which enables the ANI authentication functions.

RADIUS Server. Remote Authentication Dial-In User Service for authenticating with ANI the calling number.

Called Party. The destination called party.

Figure 6-7 ANI Authentication Application Type 1 - Call Flow



ANI Authentication Application Type 2 - Call Flow

ANI Authentication Application Type 2 enables calling subscribers to receive three authentication types based on the calling number: (1) Authentication with ANI, (2) No ANI case (if no ANI in coming packet, *Tenor AS* asks PIN number by prompt) and (3) Incoming packet has the ANI, but authentication with the ANI fails and *Tenor AS* prompts for the PIN number.

Figure 6-8 is a diagram of the call flow for ANI Authentication Application Type 2, which details the messages transmitted between the following components:

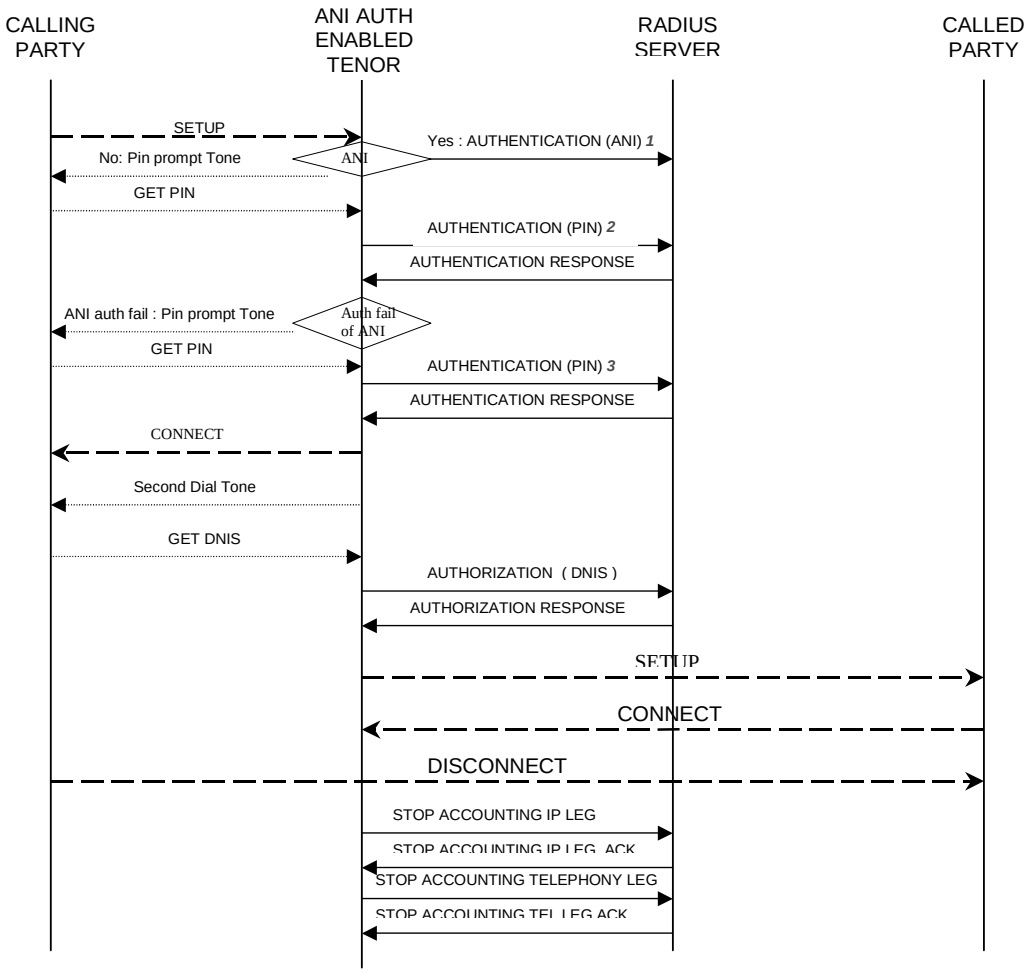
Calling Party. The originating caller using a pre-paid calling card.

ANI AUTH Enabled *Tenor AS*. The *Tenor AS* which enables the ANI authentication functions.

RADIUS Server. Remote Authentication Dial-In User Service for authenticating with ANI the calling number.

Called Party. The destination called party.

Figure 6-8 ANI Authentication Application Type 2 - Call Flow



Call Flow - Message Attributes

The sections which follow indicate the message attributes for specific message packets; the Quintum vendor ID is 6618.

Start Accounting Request Message Attributes

Table 6-3 lists the message attributes in the *Start Accounting Request* message packet sent between the *Tenor AS* and the RADIUS server. Start Accounting is optional.

Table 6-3 Start Accounting Request Message Attributes

IETF Attribute Number	Attribute Name	Vendor Specific Attribute Number	Description	Value Format	Sample
4	NAS-IP-Address		IP Address of the requesting *Tenor AS*	Numeric	3506546880
26	Cisco-NAS-Port	2	*Tenor AS*'s physical port on which the call is active	String	Fxs1/0/1
61	NAS-Port-Type		*Tenor AS*'s physical port type	String	0: Asynchronous
1	User-Name		Account number, calling party number, or blank	String	7325551212
30	Called-Station-Id		The IVR access number, normally an 800 number	String	8004609000
31	Calling-Station-Id		Calling Party Number (ANI)	String	7329701330
40	Acct-Status-Type		Account Request Type (start or stop)	Numeric	1: start, 2: stop
6	Service-Type		Type of service requested	Numeric	1: login
26	h323-gw-id	33	Name of the *Tenor AS*	String	Quintum *Tenor AS*
26	h323-conf-id	24	16 octets h323 GUID	16 octets	
26	h323-incoming-conf-id	1	A number to match multiple calls in a session	16 octets	
26	h323-call-origin	26	The call relation to the *Tenor AS*	String	Answer, Originate etc
26	h323-call-type	27	Protocol type or family used on this leg of the call	String	Telephony or VOIP
26	h323-setup-time	25	Setup time in NTP format (hours, minutes, seconds, microseconds, time_zone, day, month, day_of_month, year)	String	
44	Acct-Session-Id		A unique accounting identifier - match start & stop	String	13432
41	Acct-Delay-Time		Number of seconds tried in sending a particular record	Numeric	5

Stop Accounting Request Message Attributes

Table 6-4 lists the message attributes in the *Stop Accounting Request* message packet sent between the *Tenor AS* and the RADIUS server.

Table 6-4 Stop Accounting Request Message Attributes

IETF ATTRIBUTE NUMBER	Attribute Name	Vendor Specific Attribute Number	Description	Value Format	Sample
4	NAS-IP-Address		IP Address of the requesting *Tenor AS*	Numeric	3506546880
61	NAS-Port-Type		*Tenor AS*'s physical port type on which the call is active	Numeric	0: Asynchronous
1	User-Name		Account number or calling party number	String	7325551212
30	Called-Station-Id		Destination phone number	String	7324609000
31	Calling-Station-Id		Calling Party Number (ANI)	String	7329701330
40	Acct-Status-Type		Account Request Type (start or stop)	Numeric	1: start 2: stop
6	Service-Type		Type of service requested	Numeric	1: login
26	Cisco-NAS-Port	2	*Tenor AS*'s physical port on which the call is active	String	Fxs1/0/1
26	h323-gw-id	33	Name of the *Tenor AS*	String	Quintum *Tenor AS*
26	h323-conf-id	24	16 octets h323 GUID	16 octets	
26	h323-incoming-conf-id	1	A number to match multiple calls in a session	16 octets	
26	h323-call-origin	26	The call relation to the *Tenor AS*	String	Answer, Originate
26	h323-call-type	27	Protocol type used on this leg of the call	String	Telephony or VOIP
26	h323-setup-time	25	Setup time in NTP format	String	
26	h323-connect-time	28	Connect time in NTP format	String	
26	h323-disconnect-time	29	Disconnect time in NTP format	String	
26	h323-disconnect-cause	30	Q.931 disconnect cause code	Numeric	

26	h323-voice-quality	31	ICPIF of the voice quality	Numeric	
26	h323-remote-address	23	IP address of the remote gateway	Numeric	
44	Acct-Session-Id		A unique accounting identifier-match start & stop	String	
42	Acct-Input-Octets		Number of octets received for that call duration	Numeric	
43	Acct-Output-Octets		Number of octets sent for that call duration	Numeric	
47	Acct-Input-Packets		Number of packets received during the call	Numeric	
48	Acct-Output-Packets		Number of packets sent during the call	Numeric	
46	Acct-Session-Time		The number of seconds for which the user receives service	Numeric	
41	Acct-Delay-Time		No of seconds tried in sending a particular record	Numeric	5

Authentication Request Message Attributes

Table 6-5 lists the message attributes in the *Authentication Request* message package sent between the *Tenor AS* and the RADIUS server.

Table 6-5 Authentication Request Message Attributes

IETF ATTRIBUTE NUMBER	Attribute Name	Vendor Specific Attribute Number	Description	Value Format	Sample
4	NAS-IP-Address		IP Address of the requesting *Tenor AS*	Numeric	3506546880
61	NAS-Port-Type		*Tenor AS*'s physical port type on which the call is active	Numeric	0: Asynchronous
1	User-Name		Account number, calling party number, or blank	String	7325551212
26	h323-conf-id	24	16 octets h323 GUIDE	16 octets	
31	Calling-Station-Id		Calling Party Number (ANI)	String	7329701330
26	h323-ivr-out	1	IVR Access Number, normally an 800 number	String	ACCESS:8005551234
2	User-Password		16 octets user password	String	

Authentication Response Message Attributes

Table 6-6 lists the message attributes in the *Authentication Response* message packet sent between the *Tenor AS* and the RADIUS server.

Table 6-6 Authentication Response Message Attributes

IETF ATTRIBUTE NUMBER	Attribute Name	Vendor Specific Attribute Number	Description	Value Format	Sample
26	h323-return-code	103	The reason for failing authentication	Numeric	2 Invalid pin number
26	h323-preferred-lang	107	Lang to play prompt specified by h323-prompt-id.	ISO 639-1	En
26	h323-credit-amount	101	Amount of credit (currency) remaining in the account	Numeric	13.25

26	h323-billing-model	109	Type of billing service for a specific call.	Numeric	1:debit/prepaid
26	h323-currency-type	110	Currency for use with h323-credit-amount	ISO 4217	USD
31	h323-prompt-id	104	Index into an array of prompt files used by *Tenor AS*	Numeric	27

Authorization Request Message Attributes

Table 6-7 lists the message attributes in the *Authorization Request* message packet sent between the *Tenor AS* and the RADIUS server.

Table 6-7 Authorization Request Message Attributes

IETF ATTRIBUTE NUMBER	Attribute Name	Vendor Specific Attribute Number	Description	Value Format	Sample
4	NAS-IP-Address		IP Address of the requesting *Tenor AS*	Numeric	3506546880
61	NAS-Port-Type		*Tenor AS*'s physical port type on which the call is active	Numeric	0: Asynchronous
1	User-Name		Account number or calling party number or blank	String	7325551212
26	h323-conf-id	24	16 octets h323 GUID	16 octets	
30	Called-Station-Id		Destination phone number	String	7324609000
31	Calling-Station-Id		Calling Party Number (ANI)	String	7329701330
26	h323-ivr-out	1	Flag for subsequent authorizations in a session	String	FOLLOW_ON_FLAG:0
2	User-Password		16 octets user password		

Authorization Response Message Attributes

Table 6-8 lists the message attributes in the *Authorization Response* message packet sent between the *Tenor AS* and the RADIUS server.

Table 6-8 Authorization Response Message Attributes

IETF ATTRIBUTE NUMBER	Attribute Name	Vendor Specific Attribute Number	Description	Value Format	Sample
26	h323-return-code	103	The reason for failing authentication	numeric	2 Invalid pin number
26	h323-credit-time	102	Number of seconds for which the call is authorized	numeric	360

GLOSSARY

A

Alarm. A brief message that appears on your screen when the *Tenor AS* encounters a problem (i.e., failed interface). Alarms can be viewed through CLI (see *Command Line Interface*), a Telnet connection, or *Tenor Monitor*.

Auto Switching. If a network packet delay for an IP call becomes unacceptable, the *Tenor AS* will automatically switch the call to PSTN.

B

Border Element. Provides access into or out of an administrative domain. The *Tenor AS* has two types of Border Elements: Primary and Secondary.

Bypass Number. A telephone number that is automatically sent to the PSTN, without going VoIP.

C

CDR. Call Detail Recording. A string of data which contains call information such as call date and time, call length, calling party and called party. CDRs can be viewed through the Tenor Monitor.

CDR Server. The server (or workstation) responsible for receiving and processing CDRs as they are generated.

Command Line Interface (CLI). A configuration system you use to configure and monitor the *Tenor AS* unit via telnet connection.

Configuration Manager. A GUI interface which enables you to configure all functions in the *Tenor AS*.

Console port. RS-232 connector on System Controller card is used for connection to a PC's serial port via DB-9 null modem cable.

D

DSP. Digital Signal which provides the required signal processing for the *Tenor AS*.

E

ESD. Electrostatic Discharge occurs as a result of improperly handled electrostatic components. An ESD Antistatic Strap must be used to prevent ESD.

Ethernet. A Local Area Network (LAN) data network design that connects devices like computers, printers, and terminals. It transmits data over twisted pair or coaxial cable at speeds of 10 to 100 Mbps.

Ethernet port. A port on the Tenor AS which provides RJ-45 jacks for connection to a 10/100 Ethernet LAN switch or hub via RJ-45 cable.

Extranet. Communications with a source outside your company.

F

FXO port. Provides an RJ-11 jack for connection to the Central Office to provide direct connection to the PSTN.

FXS port. Provides an RJ-11 jack for connection to a PBX, Keyphone or phone.

G

Gatekeeper. See *H.323 Gatekeeper*.

Ground Screw. Provides earth ground to the unit if additional grounding is required.

H

H.323. A protocol standard for sending multimedia communications (i.e., voice/data) simultaneously over packet-based networks, such as IP.

H.323 Gatekeeper. An H.323 built in gatekeeper which performs IP call routing functions such as call control and administrative services to another *Tenor AS* unit or another H.323 endpoint.

Hop-off PBX Call. A toll call which is "leaked out"

of a PBX into a private network in order to eliminate toll charges.

I

Internet. A packet based network which transports voice/video/data over TCP/IP.

Intranet communication. Communication within the same company, usually through an Ethernet hub.

IP Address. A unique 32 bit address that identifies a network device is connected to the network via TCP/IP.

IVR. Interactive Voice Response enables you offer services, such as Pre-paid calling cards and Post-paid accounts to your customers.

L

LAN. Local Area Network. A local area network that carries data between workstations in the same location. Workstations in a LAN are connected together— typically by an Ethernet hub— to share information.

LEDs. Indicators as to the status of the chassis and other components of the system. LEDs appear on the front of the unit.

P

PacketSaver. A packet multiplexing technology which reduces the amount of IP bandwidth required to support multiple calls flowing between two networks.

PBX. Private Branch Exchange. Telephone switch located on a customer's premises that establishes circuits between users and the PSTN (public network).

PSTN. Public Switched Telephone Network (also known as Central Office). Telephone Company Switching facility.

R

RJ-11. A CAT 5 cable used to connect the *Tenor AS* to an FXO or FXS interface.

RADIUS. When using IVR, the RADIUS (Remote Authentication Dial-In User Service) is used for authenticating and authorizing user access to the VoIP network.

S

SelectNet™. Monitors your data network for jitter, latency, and packet loss, and transparently switches customer calls to the PSTN when required.

SIP. Signaling protocol used to establish a session on an IP network.

SNMP. Simple Network Management Protocol (SNMP) is the standard protocol used to exchange network information between different types of networks.

Subnet Mask. An IP address that determines how an IP address is divided into network and host portions according to the bits.

T

TCP/IP. Transmission Control Protocol/Internet Protocol. TCP/IP is a standard communications protocol divided into seven layers of activity. Each layer defines a different aspect of how two devices should talk to each other (i.e., when a network device should send/receive data). For example, layer one is the physical means of communications (e.g., modem), whereas layer 3 is the network type (e.g, Internet). For TCP/IP, it is a combination of two layers of communication protocol. TCP (layer 4) does the actual transport of data; IP (layer 3) sets the rules for moving the data from one end of the network to another. TCP/IP uses an IP address to identify a location for specific network devices.

Tenor Monitor. A GUI interface which provides a set of utilities to monitor the network and all system components.

W

WAN. Wide Area Network. A number of LANs connected together through a long distance communications medium. For example, your company may have a LAN in New York, a LAN in Tokyo, and a LAN in Los Angeles. When these sites connect together over the data network or

the public network, it is considered a WAN. As a result, intra-corporate information is passed through the data network from one LAN to another LAN site in a remote location.

Zone. A group of endpoints (e.g, gateways, terminals, etc.) in one corporate site.

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QUINTUM TECHNOLOGIES, INC.
LIMITED WARRANTY AGREEMENT

Quintum Limited Warranty

QUINTUM WARRANTY: Quintum warrants that under normal use and conditions (i) the Quintum hardware products covered by this warranty, for a period of two years, and (ii) all software media, also for a period of two years, will be free from significant defects in materials and workmanship from the date of purchase from Quintum or Quintum's authorized reseller or distributor (the "Warranty Period").

SERVICES:
In the event that you believe that you have discovered any such defect during one of the Warranty Periods listed above, you must call the Technical Assistance Center (TAC) at 877-435-7553 within the United States or 732-460-9399 Internationally, 9:00 AM to 5:30 PM, Eastern Standard Time, for initial problem diagnosis. Quintum Technologies will perform warranty service at Quintum Technologies designated facility, provided the customer returns the Quintum Technologies Product in accordance with Quintum Technologies' shipping instructions. Quintum Technologies' sole responsibility under this warranty shall be, at Quintum Technologies' option, to either repair or replace the Quintum Technologies Product within 10 days. All defective Quintum Technologies Products, or defective components thereof, returned under this warranty shall become Quintum Technologies' property. If Quintum Technologies determines that the original Quintum Technologies Product did not contain a Material Defect, Purchaser shall pay Quintum Technologies all costs of handling, transportation, and repairs at Quintum Technologies' prevailing rates, including all costs of providing an interim Quintum Technologies Product.

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1. Notify Quintum Technical Assistance Center on Telephone: 877-435-7553 within the United States, 732-460-9399 Internationally, Monday through Friday from 8:30am till 5:30pm U.S. Eastern time.

2. Provide Customer Services Department the following information:

    - Customer Name and Contact Name
    - Product Part number(s)
    - Product serial numbers
    - Quantity to be returned
    - Type of return (i.e., warranty return)
    - Reason for return
    - Proof of purchase (invoice or PO)

3. An RMA number will be assigned for each shipment and that number must be quoted in all correspondence relating to the RMA in question.

4. Shipment Instructions: Customer must follow any instructions supplied by the Customer Service Representative concerning where the Product is to be returned, how the Product is to be packaged, which carrier is to be used, who should pay for the shipment and any labels to be put on the package. Unless otherwise directed by Quintum's Customer Services Representative, please return product to Quintum at:

    REF RMA Number
    Quintum Technologies, Inc.
    71 James Way
    Eatontown, NJ 07724 USA

5. Following all directions given by Customer Services Representative return the Product to the address given by the Customer Services Representative quoting the RMA number.

6. Any product that is deemed failing under this Warranty and a replacement product has been shipped to the customer, the failing product must be returned and delivered to the address given by the Customer Services Representative within 30 days of the replacement being shipped.

PLEASE NOTE: All shipments require an authorized RMA number.

If the Customer does not comply with this procedure as set out above, Quintum reserves the right to charge Customer for the cost of the replacement Product and/or freight (including duties and taxes) from Quintum regardless of the reason for the return. Quintum also reserves the right to invoice the Customer for a replacement Product at the same time as the replacement is cross-shipped. This invoice will, of course, be canceled if the original Product is returned within 30 days of cross-shipment and if found to be a valid warranty return.

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Agency Approvals

EMI/EMC Standards
FCC Part 15 Class B
ICES-003
EN55022:98
EN55024:98
EN61000-3-2 :95
EN61000-3-3:95
AS/NZS 3548:1995

Safety Standards
UL60950-1
CSA C22.2 No.60950-1
EN60950:99
TS001:1970

Telecom Standards
FCC Part 68
CS-03
AS/NZ 3260:1997
AS/ACIFS002:2001
AS/ACIF S003:2001

European Directives
EMC Directive, 89/336/EEC
Low Voltage Directive, 73/23/EEC
R&TTE Directive, 99/5/EC

FCC WARNINGS

This equipment has been tested and found to comply with the limits for Class A digital device, pursuant to Part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference in a residential installation. This equipment generates, uses and can radiate radio frequency energy, and, if not installed and used in accordance with the instructions, may cause harmful interference to radio communications. However, there is no guarantee that interface will not occur in a particular installations. If this equipment does cause harmful interference to radio or television reception, which can be determined by turning the equipment off and on, the user is encouraged to try to correct the interference by one or more of the following measures:

· Reorient or relocate the receiving antenna.
· Increase the separation between the equipment and receiver.
· Connect the equipment into an outlet on a circuit different from that to which the receiver is connected.
· Consult the dealer or an experienced radio/TV technician for help.

CAUTION: Changes or modifications not expressly approved by the party responsible for compliance could void the user's authority to operate the equipment.

This device complies with Part 68 of the FCC Rules and the requirements adopted by ACTA. On the back of this equipment is a label that contains, among their information, a product identifier in the format US:6LCPF01AAS-SERIES. If requested, this information must be provided by the Telephone Company.

The REN (Ringer Equivalence Number) is used to determine the number of devices that may be connected to a telephone line. Excessive RENs on a telephone line may result in the devices not ringing in response to an incoming call. In most but not all areas, the sum of RENs should not exceed five (5.0). To be certain of the number of devices that may be connected to a line, as determined by the total RENs, contact the local telephone company. For products approved after July 23, 2001, the REN for this product is part of the product identifier that has the format US:AAAEQ##TXXXX. The digits represented by ## are the REN without a decimal point (e.g., 03 is a REN of 0.3). For earlier products, the REN is separately shown on the label.

Facility Interface Codes For Analog Services supported:

- 02LS2

Service Order Codes For Analog Services supported:

- 9.0F Full protection to the network from systems using live voice. Only approved terminal equipment can be connected to station ports

An FCC compliant telephone cord with a modular plug is provided with this equipment. This device connects to the telephone network via an RJ-11 plug and jack. The plug and jack also comply with FCC part 68 rules.

If this device causes harm to the telephone network, the telephone company will notify you in advance that temporary discontinuance of service may be required. But, if advance notice is not practical, the Telephone Company will notify the customer as soon as possible. Also, you will be advised of your right to file a complaint with the FCC if you believe it is necessary.

The Telephone Company may make changes in its facilities, equipment, operations, or procedures that could effect the operation of the equipment. If this happens, the Telephone Company will provide advance notice in order for you to make necessary modifications to maintain uninterrupted service.

If trouble is experienced with this device, for repair and warranty information, please refer to the Technical Support insert for repair information and the warranty section of this Getting Started Guide for warranty information.

In the event of device malfunction, all repairs should be performed by Quintum Technologies, Inc. or an authorized agent. It is the responsibility of users requiring service to report the need for service to Quintum Technologies or to one of our authorized agents. In the event service is required, refer to the Technical Support insert for information.

If the device is causing harm to the telephone network, the telephone company may request that you disconnect the equipment until the problem is resolved.

This registered device is capable of providing users access to interstate providers of operator services through those of equal access codes.

U.S. Service Center Information:

Quintum Technologies, Inc.
71 James Way
Eatontown, NJ 07724 USA

Canadian Notice

The Industry Canada label identifies certified equipment. This certification means that the equipment meets certain telecommunications network protective, operation, and safety requirements. The Department does not guarantee the equipment will operate to the users' satisfaction.

Before installing this equipment, users should ensure that it is permissible to be connected to the facilities of the local Telecommunications Company. The equipment must also be installed using an acceptable method of connection. In some cases, the inside wiring associated with a single-line individual service may be extended by means of a certified connector assembly. The customer should be aware that compliance with the above conditions may not prevent degradation of service in some situations.

Repairs to certified equipment should be made by an authorized Canadian maintenance facility designated by the supplier. Any repairs or alterations made by the user to this equipment, or equipment malfunctions, may give the telecommunications company cause to request the user to disconnect the equipment.

Users should ensure for their own protection that the electrical ground connections of the power utility, telephone lines, and internal metallic water pipe system, if present, are connected together. This precaution may be particularly important in rural areas.

CAUTION: Users should not attempt to make electrical ground connections by themselves, but should contact the appropriate inspection authority or an electrician, as appropriate.

QUINTUM™
TECHNOLOGIES, INC.

DECLARATION OF CONFORMITY

Application of Council Directives(s)

89/336/EEC, 93/68/ECC EMC Directives
73/23/EEC, 96/68/ECC Low Voltage Directives
99/5/EC, RTTE Directive

Standards to which Conformity is Declared:

EN55022:98, EN55024:98, AS/NZS 3548:1995
EN60950:92 +A1:92+A2:93+A3:95+A4:96
EN61000-3-2 :95, EN 61000-3-3:95
FCC Part 68, FCC Part 15 Class A
CS-03, ICES-003 Issue 3,
TS001:1970, AS/NZ 3260:1997
AS/ACIFS002:2001,AS/ACIF S003:2001

Manufacturer: Quintum Technologies Inc.

Quintum Technologies Inc.
71 James Way
Eatontown NJ
USA

Netvox Technology Co., Ltd.
No. 2, Xin Feng 2 Rd., Xiamen
Torch Hi-Tech Industrial
Development Zone,
Xiamen, P.R.C.

Delta Networks Inc.
252, Shang Ying Road,
Kuei San
Taoyuan Hsien, 333
Taiwan, R.O.C

Type of Equipment: Analog VoIP Gateway
Model Number: Tenor AS Series

We, the undersigned, hereby declare that the equipment specified above conforms to the above Directive(s) and standard(s) as of this date.

Place: Eatontown, NJ, USA

Date: 2/4/2004

Karl V. Stahl III
EMC/Product Safety Engineer

William J. Truex
Director of Operations

Technical File available through: Quintum Technologies Inc.
71 James Way
Eatontown, NJ 07724
USA